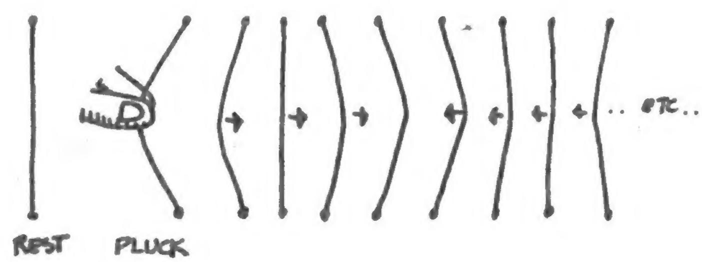
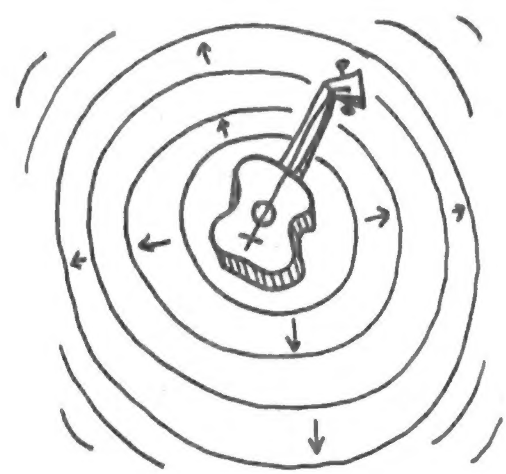
ELECTRONIC MUSIC THEORY

SOUND

Sounds are vibrations of the air caused by vibrating objects. Take a simple musical example--the string on a guitar. When it is plucked, it is pulled in one direction and released. Because it was under tension from the pulling, it snaps back to its original position and because of its momentum, it keeps going through its at-rest position to an opposite state of tension.



It proceeds to move back and forth, each time with a little less power, until it comes to rest in its original position. Almost all struck or plucked instruments vibrate in some variation of this action. When the string is released it pushes the air in front of it causing a slight extra compression of the air molecules or, put another way, a slightly higher pressure. This is called "compression". When the string flicks back, it causes a slight vacuum, or low pressure area. This is called "rarefaction". As the string vibrates back and forth more and more of these compression and rarefaction areas are created. They act like ripples in a pond, spreading out quickly and always at the same speed, the speed of sound.



If you are standing some distance away from the vibrating string when these ripples reach you, if there were some way of counting how many waves occur per second, many things could be told about the string itself! For one thing, because the speed of sound is constant, you would know how many times the string vibrates in a second. This number is called the FREQUENCY of a sound. The second thing you would want to determine is how strong the ripples are, that is, how compressed the compression wave is and how vacuous the rarefaction wave is. This strength is called the AMPLITUDE of the wave. When working with sound it can also be called the VOLUME or LOUDNESS of the sound. The Amplitude can tell you one or both of two things: how strong the source of vibrations was (i.e. how powerfully it could push air around) and/or how far away the source of the vibration is, because the amplitude of the ripples decreases with distance.

There are a number of other things we wish to detect about the sound waves that reach us. No object vibrates simply. Each has a characteristic "waveform" that, when perceived, can identify that object. This is called the TIMBRE or quality of the sound and is how we can distinguish a piano from a violin. We would want to detect these variations and have a sense of where the sound is coming from.

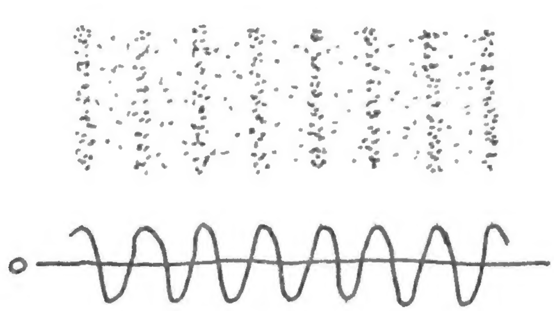
We perceive these complex waves with our ears. We hear different Frequencies as different PITCHES and we can hear them over the range of about 20 to 20,000 cycles (vibrations) per second.

We perceive Amplitude as loudness, and remarkably, we can sense the amplitudes of rustling leaves or those of a jet plane. The jet produces compressions and rarefactions nearly one million times greater than the leaves!

Without going into much detail, this is the way the ear works: The pressure inside the human head remains constant (though adjusted to the normal pressure of the atmosphere of the air.) When there are no sound waves in the air, the eardrum is at rest between two areas of equal pressure. However, when a sound wave ripples past, with its fluctuating bands of high and low pressure, the eardrum is pulled slightly outward during a rarefaction wave and pushed slightly in by the high pressure part of the wave. This means that the eardrum is going in and out at the same rate (with the same frequency) as the original sound source. The eardrum's vibrations are transmitted by means of small bones to the cochlea, a spiral organ in the inner ear filled with a liquid and coated on the inside with millions of small hairs. Each of these hairs is connected to a nerve ending through which these signals are sent to the brain.

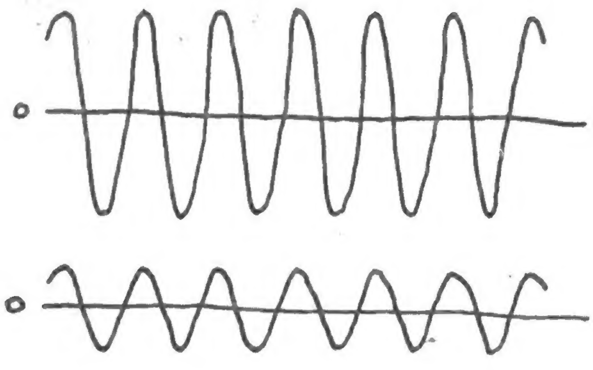
If we could take a picture of a small section of air through which a sound wave is moving, it might look like this:

In this drawing each dot represents a few million air molecules, but even with this simplification it is a rather clumsy way of describing how a wave "looks". Here is a better way is to describe the "pressure" at each point of such a wave:

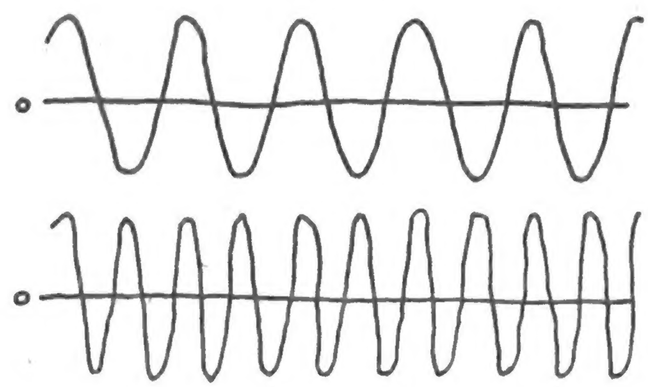


The line labelled "0" is normal pressure and the wavy line is a graph of the pressure of the wave. When the wavy line is above the 0 line, the pressure is greater than normal air pressure, when below the 0 line, it is less than air pressure.

Below are two sound waves drawn using pressure graphs:

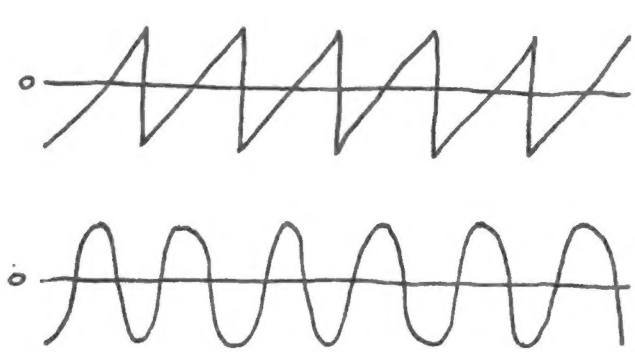


The difference between these two waves is that the top one goes further above and below the 0 line than the bottom wave. This indicates that its Amplitude or loudness is greater and is measured from "peak to peak", from the top of the highest peak to the bottom of the lowest trough. Below are two more waves.



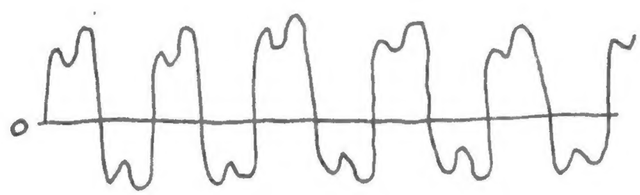
Notice that in this case the amplitude of the two waves is the same, but that in the same length of time there are twice as many excursions up and down in the bottom wave as in the top -- that is the bottom wave has twice the frequency of the top wave. The bottom wave will sound ONE OCTAVE HIGHER than the upper wave. If a wave has twice the frequency of another wave, we hear it as one octave higher. Notice that if the first octave starts out at 80 cycles per second (or 80 Hertz which means the same thing), then the next octave starts at 160 Hertz (twice the first), the third will start at 320 Hertz, the next at 640 Hertz, then 1280, 2560, and 5120 Hertz. Whereas the first octave had a range of only 160 cycles per second, the top octave had a range of 2560 cycles per second! But to our ear/brain both sound like a single octave.

Below are two waves:



These are two wave types that you will find on most synthesizers; the top one being a SAWTOOTH wave and the bottom being a SINE wave. The two waves in this drawing both have the same frequency and the same amplitude but a different SHAPE. The shape of a wave affects its TIMBRE or sound quality. Picture your eardrum being pulled in and out by the two waves shown above to see the difference in the kind of motion the liquid in the cochlea would have. In the real world, of course, nothing can vibrate in quite these shapes and if it could, the air cannot ripple in quite this fashion and if it could the eardrum cannot be moved in precisely this way. But it can all come remarkably close.

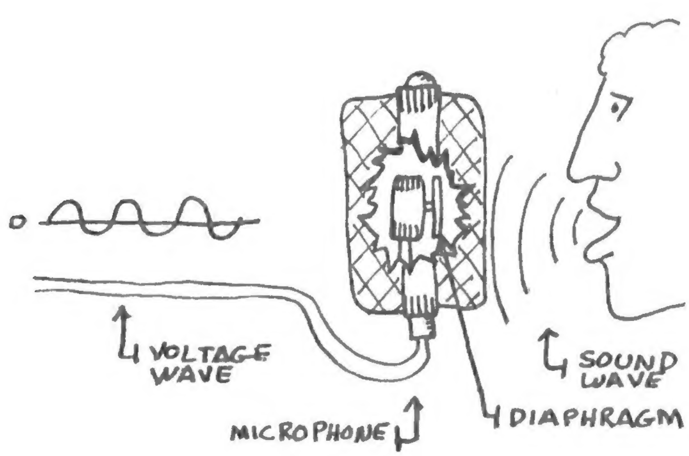
Below is what a guitar sound wave might look like:



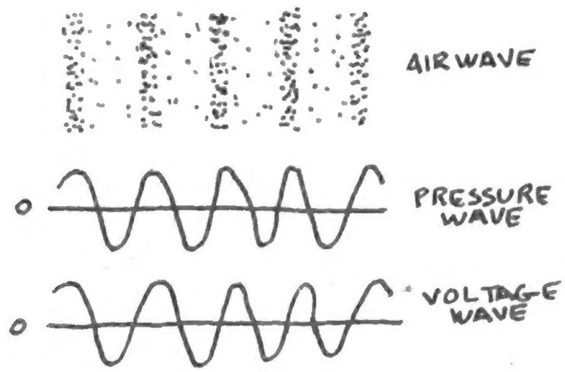
VOLTAGE

Voltage can be considered to be electric pressure. By the middle of the 19th Century, many of the advantages of converting sound waves (rapidly changing atmosperic pressure) into voltage were discerned. Primary among them was that while sound waves died out relatively rapidly, voltage waves could be sent thousands of miles over wires, around corners and through walls,. The main problem was how to convert sound waves into voltage waves and then, after a journey of perhaps a hundred miles, convert the voltage waves more or less accurately, back into sound waves. In other words, the problem was the invention of the telephone.

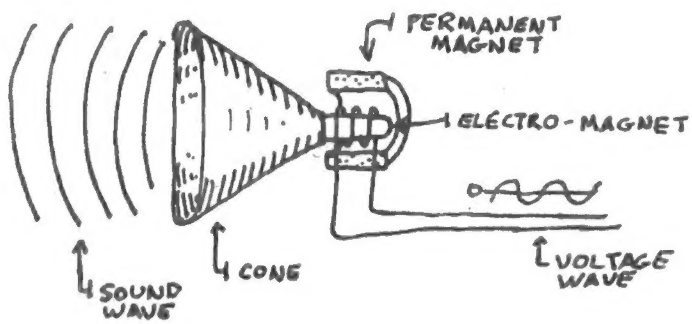
A Microphone is a device for converting sound waves into voltage waves, or atmospheric pressure into electric pressure. The simplest microphones have a diaphragm which acts much like the eardrum in its response to sound waves. It is pushed inwards by a compression wave and pulled outward by a rarefaction wave. This diaphragm is attached to a device which, when it is pushed inward creates a Positive Voltage and when it is pulled outward creates a Negative voltage. When the diaphragm is at rest, its output is Ground-- or 0 volts.



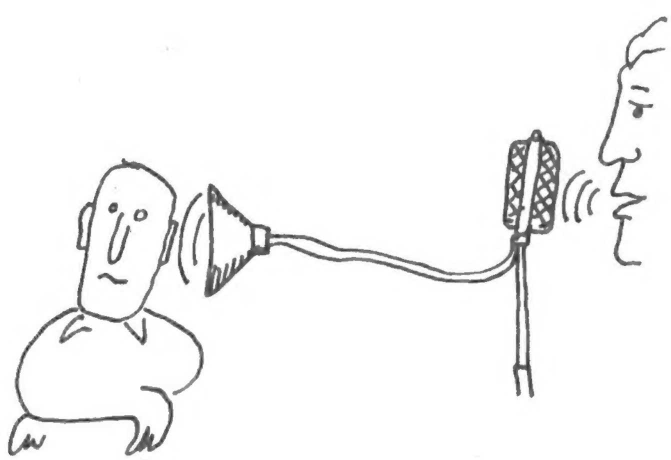
Because of this one-to-one correspondence the voltage output of a microphone is said to be isomorphic with the sound wave input.



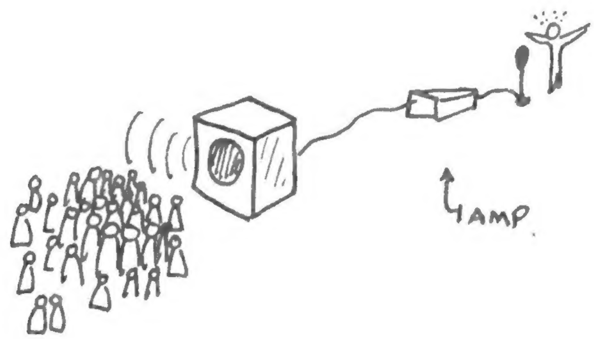
A Speaker is a device that takes a voltage wave and converts it into a sound wave. Though there are many kinds of speakers the most common ones work by moving a cardboard speaker "cone" with an electro-magnet.



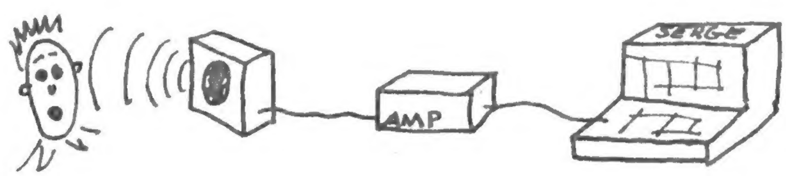
In this kind of speaker the coil of wire attached to the speaker cone sets up magnetic fields which push and pull itself in and out from the permanent magnet as the voltage changes, thereby pushing and pulling the cone in and out. This creates rarefaction and compression waves in front of the cone. The speaker cone, therefore, reproduces the movement of the diaphragm of the microphone and in so doing reproduces the original sound wave.



It was the ability of such a system to transmit sound over long distances that first attracted attention. It soon became clear that there were other advantages. Once the sound wave was converted into a voltage wave, it was far more malleable. It could be amplified, for instance, so that when the speaker re-created the sound it could be louder than the sound originally picked up by the microphone.

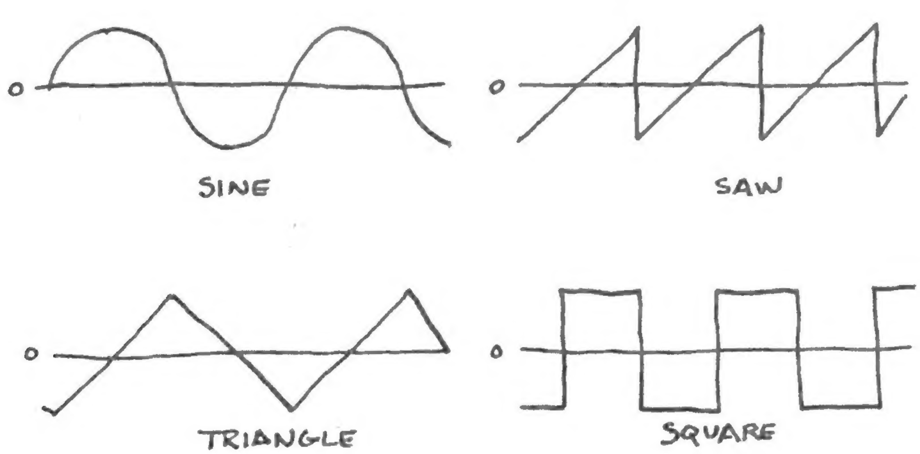


A speaker doesn't know where the voltages it is receiving are coming from. Its cone will move in response to any varying voltage. A SYNTHESIZER is a device which creates and sculpts voltages of various shapes that, when directed to a speaker, create sound that can be used in musical settings.



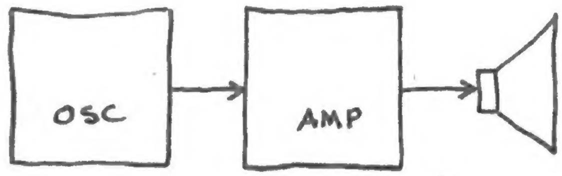
THE DEVELOPMENT OF THE SYNTHESIZER

From the earliest days of electronics, there have been various devices to create and alter voltages of audio frequency. We've already discussed the amplifier which takes an input of a varying voltage and puts out that same varying voltage magnified in amplitude. Another device is the oscillator, which simply puts out a varying voltage in a number of simple shapes:

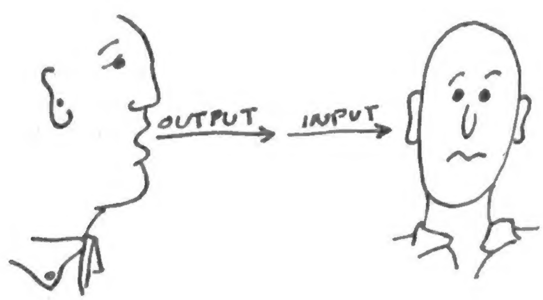


A knob, or POT (short for POTentiometer, which is the device the knob turns) on the front of the oscillator would determine the frequency of these waves, that is, how often in one second the wave would rise and fall.

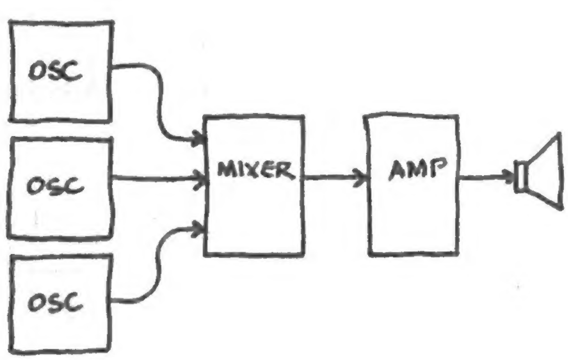
The first step towards electronic music was taken when the OUTPUT of the oscillator was connected, or PATCHED to the INPUT of the amplifier. The OUTPUT of the amplifier was sent to the speaker.



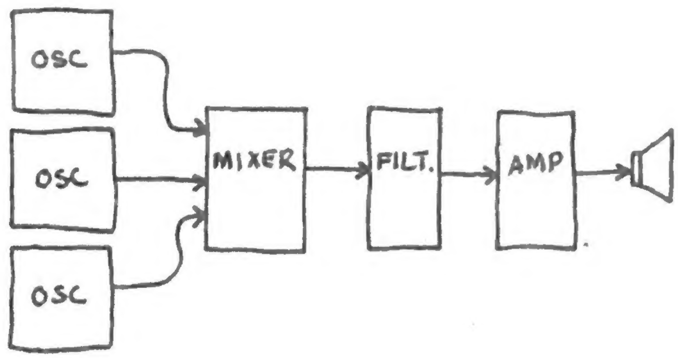
Note the "block diagram" used above. In this form of notation a block indicates an electronic device. The arrow coming out of a device is its output, while an arrow going into a device is its input. An output of one device is always the input to another device. The output of a speaker goes to the input of your ear. What outputs from your mouth inputs into someone else's ear.



Another device was the Mixer, which takes inputs and adds them together to produce a single output. Unlike the amplifier the mixer has more than one input.



Still another important device was the Filter. A filter is a device that can eliminate or accentuate various frequency components of a complex sound. For instance it can be used to eliminate all the very high components (the hiss) in a sound, by only allowing those frequencies in the range of the human voice to pass. A pot on the front of the filter controls which frequencies will be attenuated or eliminated.



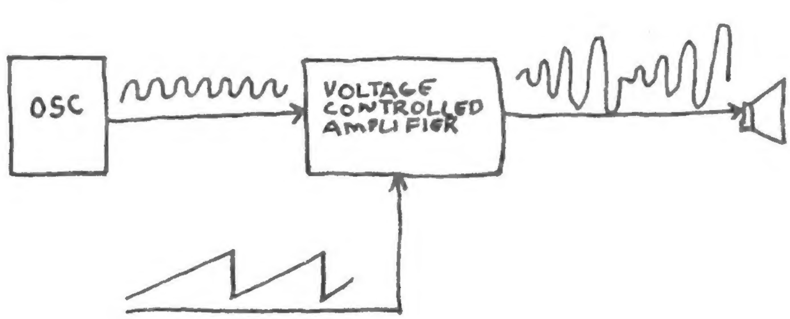
There were two problems with this procedure of adding device after device. The first was that very quickly there were just physically too many knobs to twiddle. The second problem was that the knobs couldn't be turned quickly or precisely enough. The amplifier could not be turned up and then quickly down again fast enough to make the "sound envelope" of a single whack of a drum.

The invention of the tape recorder, just after World War II, solved some of these problems. A single sound could be produced electronically, recorded on to a short piece of tape, and spliced onto another previously made sound and so on until a string of sounds had been made. Two of these tapes could be mixed together through a mixer and recorded on a third tape. The speed of the tape machines could be varied, and the segments could be reversed or even cut to form spliced "envelopes". This was (and still is) a very tedious process, but it is a very rich and flexible one. A studio built to be able to produce electronic tapes in this way is called a Classical Electronic music studio.

The first major improvement in the classical studio came from Columbia University where they devised a controller which could set all the dials instantaneously from the instructions given on a punched paper tape.

It wasn't until the Sixties that the synthesizer as we now know it was designed by Don Buchla and Robert Moog by adding Voltage Control to the classical studio.

The Voltage Controlled Electronic Music Synthesizer solved both of the two major problems of the classical studio by employing Voltage Control which works in the following manner: Each device is given a special input called a Voltage Control Input. This input accepts a voltage such that as this voltage INCREASES it is JUST LIKE TURNING UP THE KNOB ON THE FRONT OF THE DEVICE. And when the voltage goes down it is like turning down the pot on the front of the device. That is, a voltage can be used to CONTROL the device. For instance, in a voltage controlled amplifier, if the voltage at the voltage control input increases, it turns the amplifier up and makes its output louder.

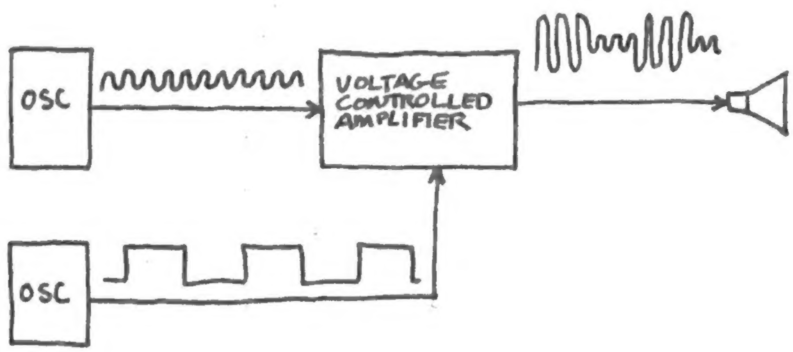


(Note that in these block diagrams, as a matter of convention, the control voltage input is on the bottom of the device, the "signal" input is on the left side and the output is on the right side.)

In a voltage controlled oscillator, a rising voltage at its voltage control input would make its frequency rise. For each device the control voltage affects only the function of that device.

Control voltages solved both problems of the classical music studio: With enough control voltages you could change all the settings of all your devices. And secondly you could change these settings so rapidly as to seem instantaneous. You could change the settings very, very slowly, or you could change them at audio frequencies, for instance 500 times per second. When a devices's settings are changed at those rates, some very strange things begin to happen, many of which can be musical.

The only problem left, of course, is where to get all these control voltages. This problem is not as great as it seems for a control voltage is identical to any other kind of voltage. For instance, we could use an oscillator to control an amplifier since the output of an oscillator is a voltage!



In the above example Oscillator #2 is controlling the amplifier, making the signal from Oscillator #1 louder and softer.

Most of the early synthesizers have two different sets of patch cords, one for the control voltages and one for the signals, even though the voltages themselves are indistinguishable. The Serge System does not make this distinction.

THE SERGE SYSTEM

The SERGE SYNTHESIZER is a Voltage Controlled Modular Music Synthesizer. By MODULAR it is meant that it is composed of separate devices or modules which must be patched together to produce a complex sound. By Voltage Controlled is meant that almost all of these devices can be controlled by a voltage as well as by their own pots. By music is meant that the Serge can be used to create complex, ordered sound, and by Synthesizer is meant that it needs no other input (though it is able to accept one) and that it can create, or synthesize, sound.

There are Four basic kinds of Modules on the Serge. Many modules can serve more than one of these functions:

SOUND SOURCES. The basic sound source is the oscillator though there are others such as white noise. Sounds from the external world, so long as they have been converted into appropriate voltages (by the use of microphones or pickups) can also be used as sound sources. Oscillators are completely voltage controllable.

SOUND PROCESSORS. Processors are devices that input one or more signals, operate on these signals, and then output a different but related signal. Mixers, filters, wave shapers, amplifiers are all processors. Almost all of these devices are voltage controllable.

CONTROL VOLTAGE SOURCES. Control voltage sources are devices that are used to create the voltages which are used to control other devices. The keyboard, for instance, puts out a voltage which can be used to control the setting of an oscillator. Other devices are envelope generators, sequencers, sample/hold devices and envelope followers. These devices are voltage controllable themselves, making possible complex levels of control.

CONTROL VOLTAGE PROCESSORS. These devices input a control voltage, operate on it, and output a related but different voltage. Processors and portamentoes are examples of these modules.

Each module on the Serge is surrounded by a border with the name of the device at the top and the Serge logo at the bottom. In some cases there is more than one device in a module and these are referred to as "dual" or "triple" modules. These dual or triple modules are two or three completely separate, though functionally identical modules.

Every module has at least one output. Outputs are usually enclosed within a border of their own.

All processor type modules have at least one input as well as an output.

Most modules have control voltage inputs which control the function of the module. These inputs are of two basic types:

PROCESSED INPUTS which have a pot associated with the input jack that can attenuate, amplify and/or invert the control voltage.

UNPROCESSED CONTROL VOLTAGE INPUTS affect the given module in a predetermined way.

Most modules have one or more pots that can control the function of the module without a control voltage. In most modules these pots control the basic setting of the module on which the control voltages operate. These pots are labelled with their function, for instance. GAIN, FREQ, etc.

The jacks on the Serge are color coded.

BLACK JACKS indicate audio or signal voltages.

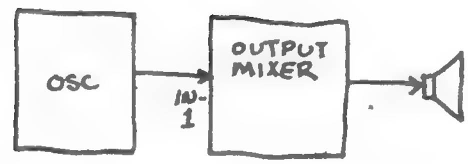
BLUE JACKS indicate control voltages.

RED JACKS are pulse input/output jacks and are used to turn modules on and off, have them step through stages and control the timing of various functions.

OTHER COLOR JACKS are special jacks whose function will be described individually.

LEARNING PATCH NUMBER TWO

STEP ONE

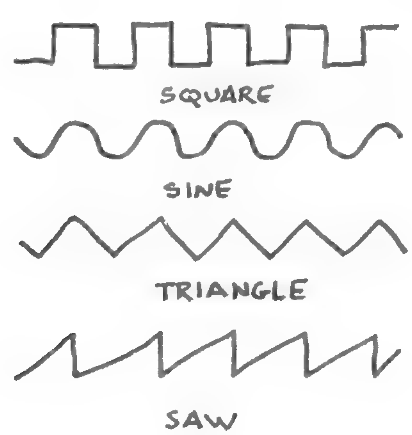


(Note: The output mixer should be set the same as in the First Learning Patch. In this section, block diagrams will be used to represent the patches. Each module is represented as a block. Its signal output is from the right side of the block. Signal inputs are shown going in to the left side of the block. Control voltage inputs go in to the bottom of the block, and control voltage outputs are shown coming off the top of the block. Each of these inputs/outputs will be labelled on the diagram. Any special pot settings necessary to make the patch work will be listed below the diagram. On some of the diagrams drawings of the waveforms will be drawn next to the appropriate patchcord.)

There are two basic oscillators in the Serge system: the New Timbral Oscillator and the Precision VCO. They are identical oscillators except for some control and output functions unique to each. This discussion will concentrate on the New Timbral Oscillator but you can try it with both the oscillators.

1.1 Set up the above Patch on your Serge. The SINE out of the OSC (the abbreviation "OSC" will be used from now on to refer to any oscillator, either a New Timbral Oscillator or Precision Controlled VCO) should be patched to Input #1 of the Output Mixer.

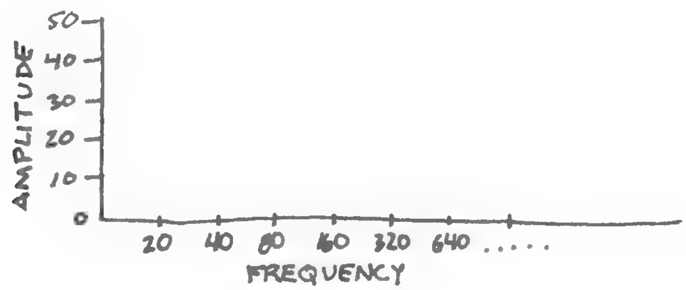
1.2 OSCs produce repetitive varying voltages referred to as "waves". These waves are produced in different "waveshapes" of which SINE, SAW, TRIANGLE and RECTANGULAR are the most common. An OSC can produce these waveshapes at different frequencies. The frequency of a wave determines its pitch. The higher the frequency of a wave the higher its pitch. The shape of a wave determines its Timbre or sound quality. Each OSC on the Serge provides a number of simultaneous outputs, all at the same frequency but with different wave shapes.



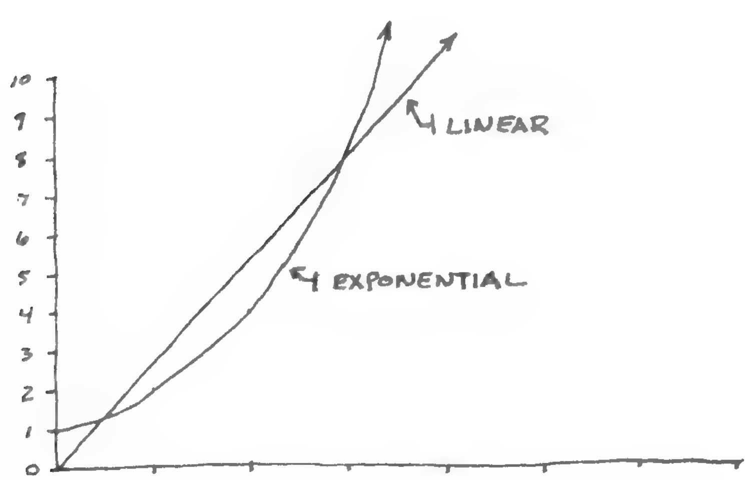
While most OSCs on most synthesizers can produce waves across the entire spectrum of human hearing -- about 20 cycles per second to 20,000 cycles per second-- (cycles per second will be referred to as Hertz), some OSCs on the Serge synthesizer, can go below this threshold. Waves of these low frequencies are useful as control voltages.

1.3 A Sine wave is the simplest form. Any waveform except a perfect SINE wave can be treated as a combination or mix of simpler waveforms. That is, ANY wave can be analysed as a mix of Sine waves of specific frequencies and amplitudes.

1.4 One way of visualizing this is with a chart that has the audible frequencies across the bottom and amplitude on the verticle axis:



Note that the scale across the bottom is EXPONENTIAL; that is, each interval marked off is TWICE the frequency of the previous interval even though the intervals are of equal lengths. This is the way we hear, with each octave having twice the frequency spread of the previous octave (e.g. 20,40,80,160,320,640...) and yet these intervals sound identical to our ears/brains. The Exponential scale contrasts with a LINEAR scale where each interval is a set distance from the previous interval. For instance a linear scale would procede 20,40,60,80,100,120,140.



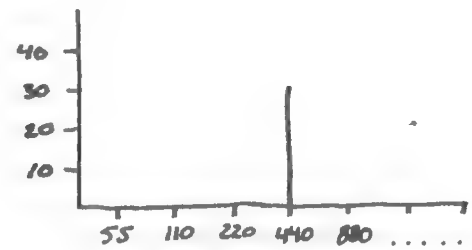
$x; 2x; 3x; 4x, 5x; 6x; 7x;$ etc.

LINEAR TERMS

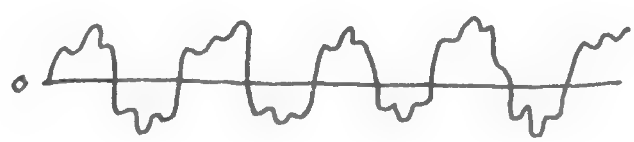
$x; x^2; x^3; x^4; x^5; x^6; x^7;$ etc.

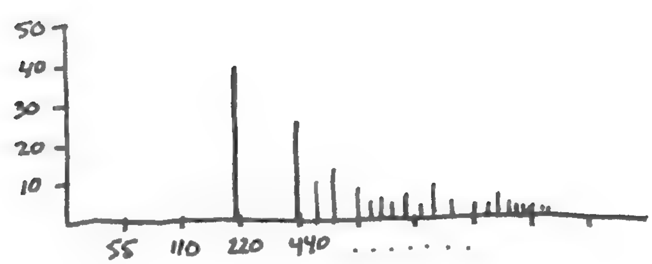
EXPONENTIAL TERMS

To notate a sound on this chart, place a vertical line at the point where each component Sine wave occurs. The height of the line will indicate the relative amplitude of the Sine wave. This vertical scale is also exponential and is measured in Decibels. Though our actual perception of loudness is not quite this simple (we are less sensitive, for instance, to frequencies at the top and bottom of the scale), generally speaking, the higher the Decibels the louder the sound. For instance a pure Sine wave with the frequency of 440 would be shown like this:



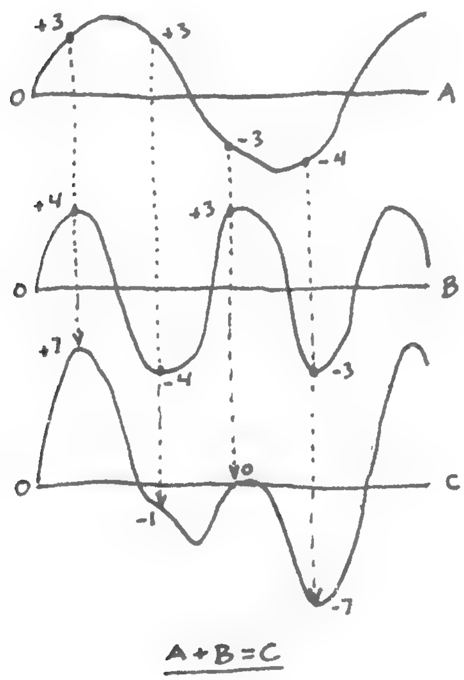
1.5 Most sounds, including electronic sounds, are composed of more than one sine wave. We now have two ways of picturing a sound,: its pressure or voltage wave and its sine-wave spectrum. The "shape" of a wave refers to its voltage as can be seen on an oscilliscope. This is called a time-domain dsiplay. The spectrum graph is called a frequency-domain graph and is an analysis of the voltage waveform. Below is a wave and its hypothetical frequency-domain spectrum.

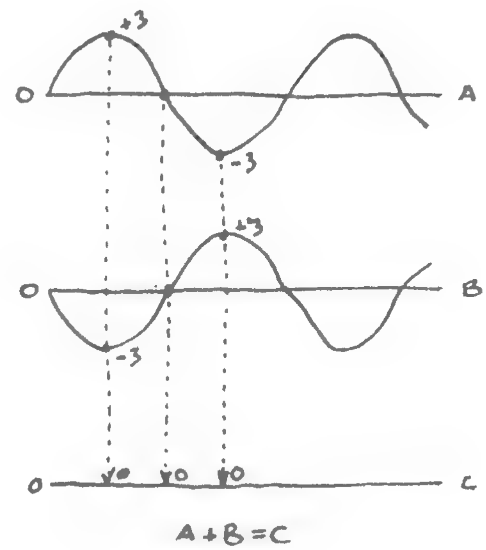




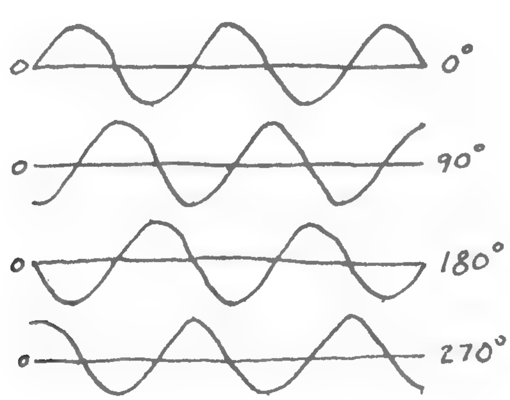
1.6 To determine the overall shape of a wave from its component sine waves, the values of the component waves AT EACH INSTANT are added together.

This also means that if two waves of identical frequency but of opposite "phase" (one goes up while the other goes down) are mixed together, silence will result.



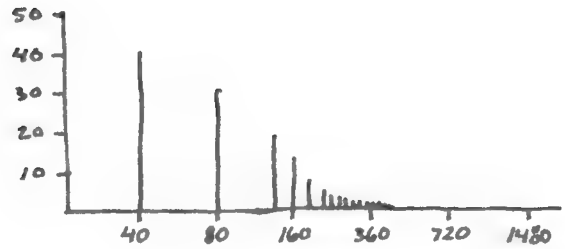


Phase is noted in degrees where 360 degrees brings a wave right back to where it started.



1.7 In theory, any sound can be created by adding together sine waves of the correct amplitude, frequency and phase. This is called "additive" synthesis or "Fourier" synthesis. This technique is of limited use in the synthesizer because the number of sine waves would have to be tremendous.

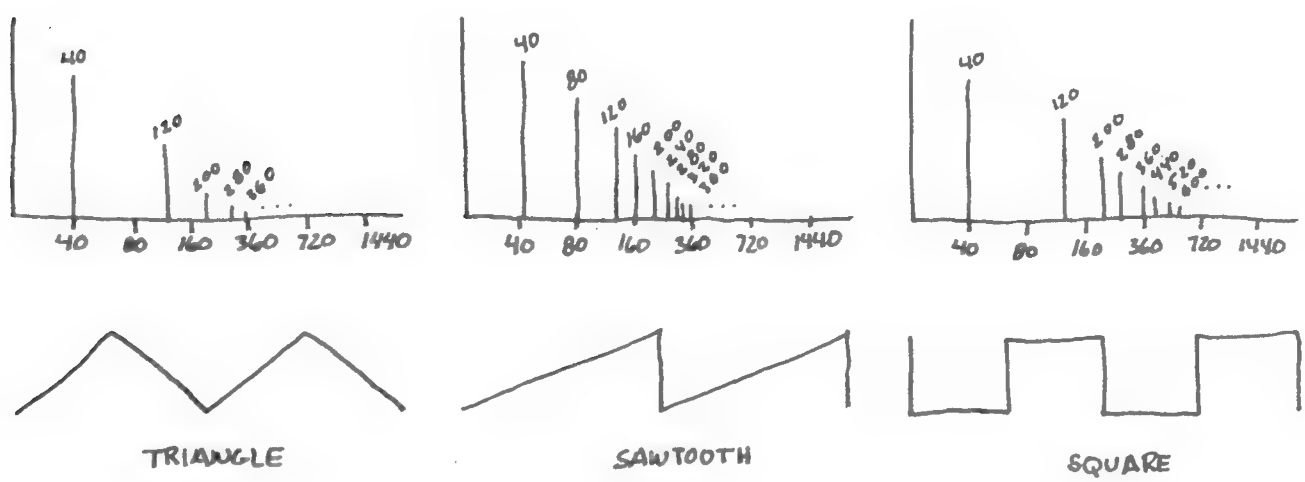
1.8 Another reason that this technique is not often used is that most sounds, and almost all musical sounds, are composed of sine waves in "harmonic" relationship to a "fundamental". The fundamental usually corresponds to the apparent pitch of a complex sound and is usually the lowest strong sine wave of the sound. If "X" is the fundamental and the other sine waves are in a harmonic relationship to it, then there is a sine wave at 2X,3X,4X,5X...etc. These sine waves are called "overtones" and they generally decrease in amplitude as they get higher in pitch. Below is the spectrum of a typical acoustic musical instrument such as a guitar:



Note that the overtones seem to be getting closer and closer together on the spectrum chart the further they get from the fundamental. We hear them in this fashion. Remember that the audio spectrum as we perceive it is exponential, but the overtone, or harmonic series, is linear!

To calculate the positions of the harmonics add the fundamental frequency to itself to get the first overtone; add it in again to the total to get the second harmonic, again to get the third and so on. An example would be: First: 100; Second: 100+100; Third 100+100+100; Fourth:100+100+100+100; etc. Thus it can be seen that the frequencies are spaced at equal, absolute spacings, that is the harmonics fall on a Linear graph.

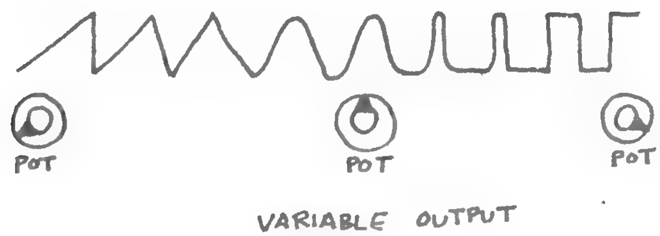
1.9 Besides the Sine wave the New Timbral Oscillator also has a Sawtooth output, a Triangle output and a Variable waveform output that can put out a Square wave, or other waveforms. (The Precision VCO has all these outputs except for the Variable waveform.) Below are the voltage or pressure diagrams of these waves and the spectrum charts of these waves.



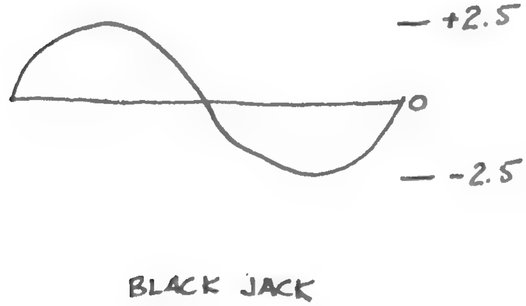
Additive synthesis can be greatly simplified by using these more complex sounds since these waves will often contain the desired harmonics.

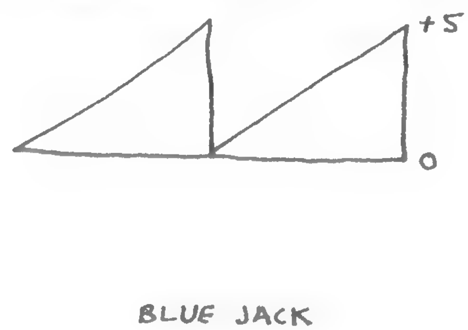
1.10 The Triangle and Square wave contain only the odd harmonics (harmonic #1,#3,#5,#7, etc), although the amplitude of these harmonics decreases more rapidly in the Triangle than in the Square. The Sawtooth wave contains both even and odd harmonics that decrease at about the same rate as in the Square wave.

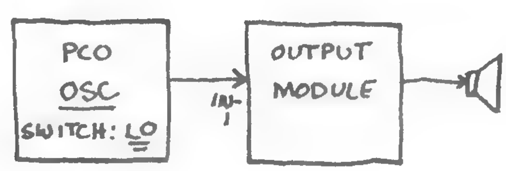
1.11 Try the different outputs of the OSC including the Variable output. The pot directly below the Variable output adjusts the shape and therefore the timbre.



1.12 Though all the outputs of the OSC are of the same amplitude, the saw and the square wave may seem louder because our ears tend to hear complex sounds as louder than pure ones. All waveforms from the oscillators have a 4 to 5 volt peak-to-peak voltage. The Sine output is from +2.5 to -2.5 volts. Black jack outputs typically have this voltage range. The other outputs of the osc. have a voltage range of 0 to 5 volts (still a 5 volt over-all amplitude).

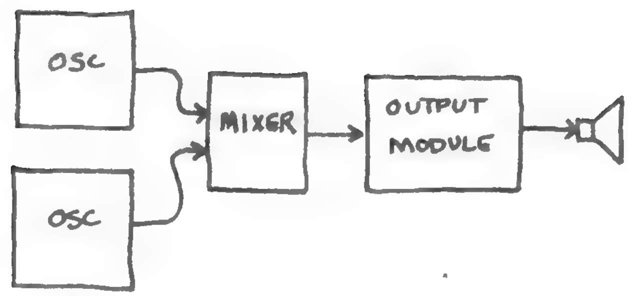






1.13 This patch uses the Precision Controlled VCO with the HI/LO switch at its LO setting to change the range of the oscillator from a range of 20 to 20,000 Hertz (audio range) to a range of .01 to 500 Hertz. Start with the Pitch pot at its furthest right position and begin slowly moving it to the left. The pitch should get lower and lower until a series of clicks appears simultaneous with the sound. The further left you go the more the pitch drops away until you are left with only clicks. Because hearing only goes down to about 20 Hertz you can no longer hear the frequency as a pitch, but the sharp edge of the sawtooth wave pulls back the speaker cone each time producing the "click". If you try the same thing with a sine wave, you will hear nothing, for there are no sharp "edges" on a sine wave. But if you can see your speaker, you will note that the cone is still moving in and out silently and slowly, responding to the changing voltage. Most audio amplifiers can only go down to a certain frequency after which there will be no motion in the speakers at all.

STEP TWO



2.1 Patch the Sine output of two OSCs to two of the inputs of the upper mixer of a Dual 3-input Audio Mixer. Note that this module is a dual module and that the top mixer is totally separate from the bottom mixer. Patch the output of this mixer to the Output Mixer (QCA, UPAP, etc.). Each of the inputs of the mixer has a Pot associated with it that can limit, or "attenuate" the gain of its input. The output of this module is the summation of all its inputs at their assigned gain.

2.2 Tune the two OSCs so that they are very close in pitch and set their gains so that they are at the same level. When they are exactly the same pitch, they should sound like a single sound. If they are a few Hertz apart, you will be able to hear a "beating" between them. The frequency of this beating is the difference in frequency between the two sine waves.

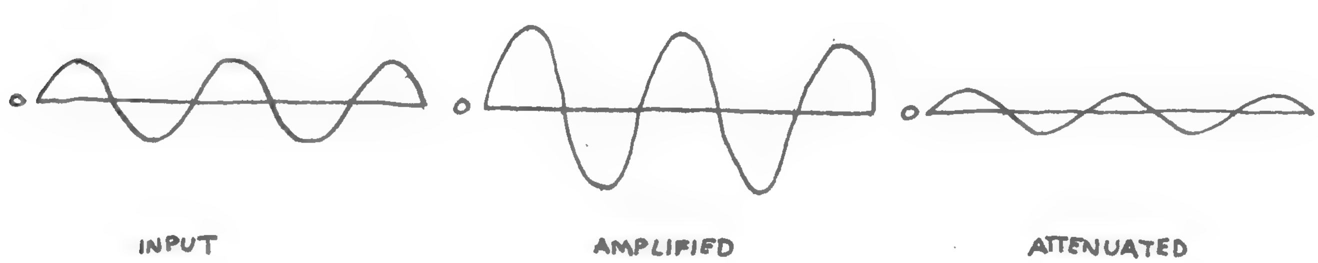
2.3 Try adding a third Sine wave to make a tri-tone.

2.4 Unpatch all but one of the OSCs. Turn up its mixer pot and note that the ear hears the sound as unchanging EXCEPT that it gets louder and louder. This is comparable to the way the eye sees a photograph and its blow-up as identical only the blow-up is larger.

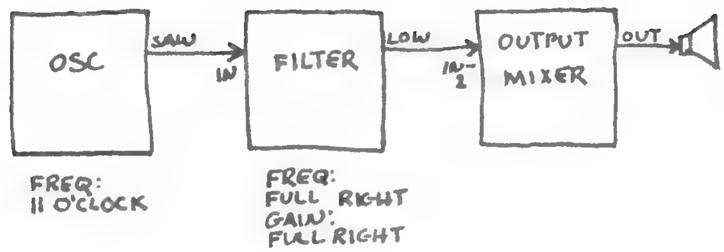
2.5 Very few sounds in the world have a steady amplitude or gain. How a sound's gain changes is one of the clues as to what is vibrating. It is one of the components of the over-all feel of a sound. For instance, a piano note gets very loud very quickly when struck, then slowly gets softer and softer. If the way this amplitude changes were altered, we would not easily recognize it as a piano sound. This amplitude shape is called the ENVELOPE of a sound, because like a letter in an envelope, the sonic information is contained within it.

2.6 Because of the way our ear/brains process sound, the amplitude of a sound must increase exponentially in order for us to perceive it as linearly increasing. For this reason the pots on the mixer are logarithmic.

2.7 When the pot of the associated input is turned to the right, the sound increases in level. Turning it to the left will cause the sound level to decrease. The shape of the wave and its frequency remain the same except for this change in amplitude:



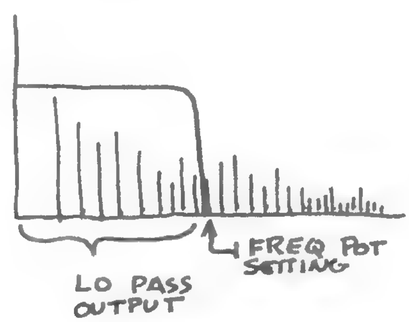
STEP THREE



3.1 A FILTER is a module which makes it possible to eliminate certain components of a sound, depending on its frequency. As we said earlier, every sound can be thought of as the summation of a number of sine waves, each with a different frequency. The Filter allows us to listen to those Sine waves in a sound which fall above, below or directly around a Frequency set by the Pot labelled "FREQ" on the filter.

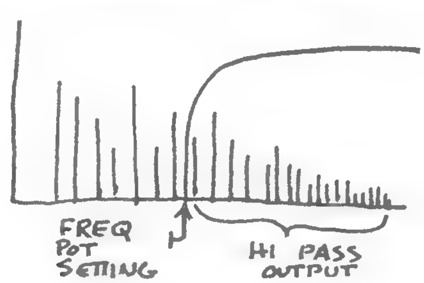
3.2 While there are a number of different outputs on the filter, all outputs can be thought of as different combinations of HI pass and LO pass outputs.

3.3 A LO PASS filter lets PASS through to the output all those sine wave components in the input sound which are LOWER than the Frequency set by the FREQ Pot.

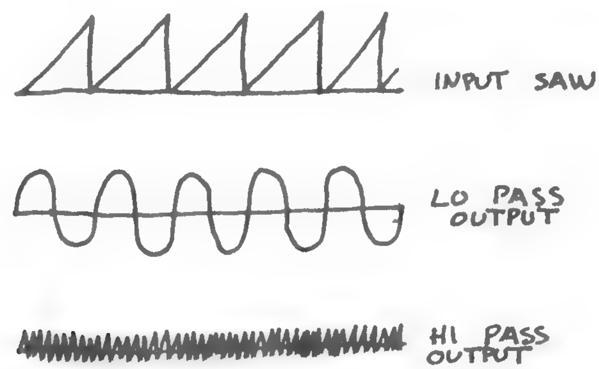


3.4 Slowly turn the FREQ. Pot to the left and the "hissy" sounds will start to disappear. As turning of the pot continues the mid-range will disappear, and finally there will be nothing left but a very low sound, the fundamental, of the oscillator. If the FREQ pot is turned even further, it will eliminate this sine wave as well, leaving no sound.

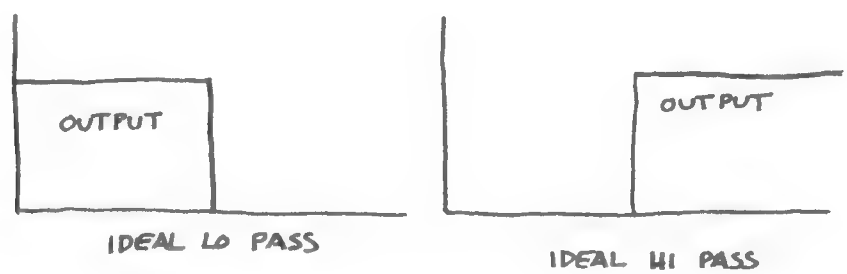
3.5 Re-patch the above patch using the HI PASS output. Now the filter lets PASS only those sounds which are higher than the frequency set by the FREQ POT. It lets pass to the output only the High frequencies of the input sound. Starting with the FREQ pot full left and slowly turning it right, the fundamental will drop out and then the mid-range. ONLY the hiss, or very top part of the spectrum, will be left of the input sound.



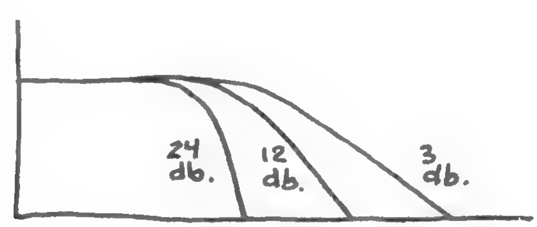
3.6 In terms of waveshape the LO pass filter SMOOTHS out a wave. It finds those components which change the least. Mathematically, it can be said to take the integral of the wave. A HI PASS filter takes the derivative of a wave. That is, the HI pass filter finds those parts of the wave which change the fastest. Below are some typical waveform outputs from HI and LO pass filters:



3.7 An "ideal" filter would not allow any sounds HIgher or LOwer than its cut-off frequency to Pass. It would look like this on a spectrum chart:



But all filters fall short of these ideals, not only because no technology is perfect but because such filters do not produce very musical sounds. The cut-off sharpness is measured in db/oct with 0 db/oct being no cut-off at all and 60 db/oct being about as sharp as we can hear. Most synthesizer filters are in the 3 to 24 db/oct range. The Variable Q Filter has a 12 db/oct cut-off. The Variable Slope VCF's cut-off can be varied from 0 to 12 db/oct using the second POT below the input labelled VC SLOPE.

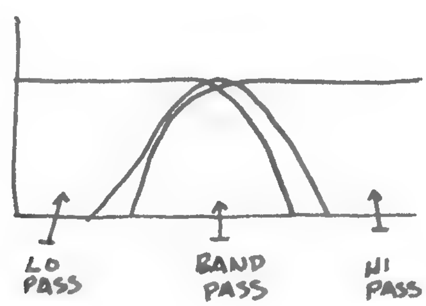


LO PASS CUT-OFFS

Another phenomenon of filtering available on the Serge is called the "Q". Most filters tend to amplify the frequencies near the the cut-off. The more these frequencies are amplified, the higher the Q of the filter. In most cases, the higher the Q, the sharper the cut-off. Knocking on the table is a typical low Q sound from the natural world. A drum head has a medium Q and a bell has a high Q.

3.8 On the Variable Q VCF the Q can be adjusted by using the POT just below the VCQ label. On the Variable Slope VCF when the slope is set so that it is very sharp (full right) the Q is very high. Using a very HI Q it is possible to "scan" through the overtones of a sound by slowly turning the FREQ Pot of the filter. Everytime the Freq. is the same as an overtone it will amplify that overtone.

3.9 The BAND output of a FILTER filters out everything but a area around the Frequency set by the Freq pot. It is useful for listening to a single part of a more complex sound. Below is a diagram of how HI, LO and BAND Pass are related to each other.

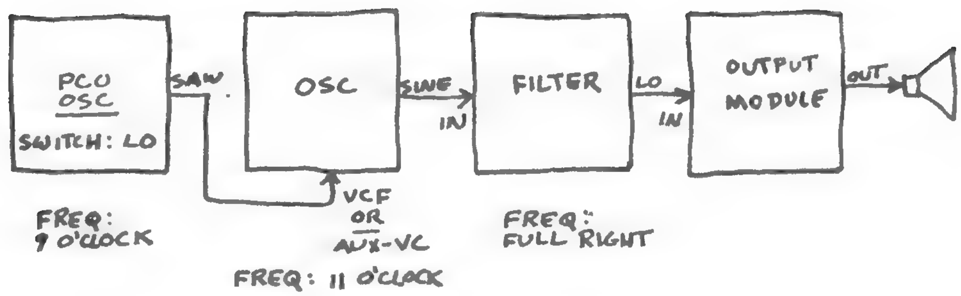


3.10 While HI pass filtering occurs only rarely in nature (a cheap transistor radio tends to be a hi-pass filter to music by cutting out the lows), LO pass filtering abounds. In many musical instruments, a piano for instance, once the string is struck the highs tend to be filtered out leaving only the lows-- the typical action of a LO pass filter. The human mouth is also a LO pass filter and is responsible for our vowel sounds, which again are LO pass filter sounds.

3.11 The GAIN pot on the Variable Q VCF controls the level of the signal input exactly like the POT on a mixer. It must be turned up to hear any output. If the Q of the filter is set high, the GAIN should usually be turned down so that when the FREQ of the filter and the frequency of an overtone coincide, the filter is not overdriven. (Sometimes this is the desired effect. Even though the filter will overload, no damage will be done.)

3.12 The Variable Slope VCF has two independent inputs which can be manually "cross-faded" or mixed together using the MIX pot. If IN-1 is used as the input jack, be sure that the MIX pot is set to the left; vice versa for IN-2.

STEP FOUR



STEPS One, Two and Three could have been been set up in a classical music studio. STEP FOUR begins the exploration of Voltage Control; a technique which extends electronic synthesis to its modern form.

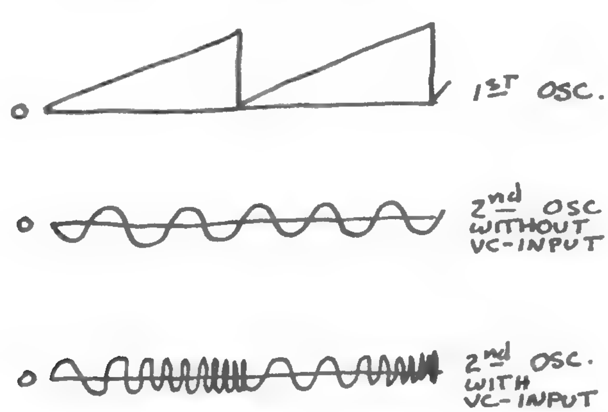
4.1 Patch the SAW output of a Precision VCO to the VC-AUX or VCF input of a second OSC, either a PCO or NTO. The PCO is used as the first OSC because it has a range switch allowing it to oscillate at very low frequencies. This range switch should be set to LO.

A SAW wave is a voltage which rises from 0 to 5 volts and then swiftly drops back to 0 volts. It does this over and over again.

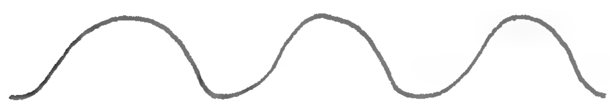
When an OSC is voltage controlled it is like TURNING its FREQ pot by REMOTE CONTROL. When this controlling frequency is rising it is exactly like turning the FREQ pot to the right. When the controlling voltage falls, it is like turning the FREQ pot to the left.

4.2 Turn the VC AUX or the VCF POT on the second OSC full right and the GAIN up on the audio mixer until the sweeping sounds of the oscillator can be heard. The sound will rise higher and higher and suddenly fall back to a very low sound only to begin rising again. The pitch is produced by the second oscillator. The first oscillator's SAWtooth wave is causing it to rise and then swiftly fall.

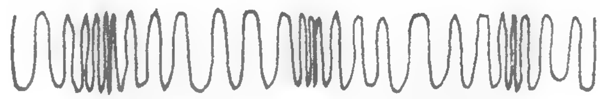
This is a stylised picture of the pressure wave being produced:



If the TRIANGLE or SINE output of the first OSC are used instead of the SAW, the following waveshapes are produced. These can be heard as different patterns of rising and falling pitches.



SINE WAVE MODULATING SINE WAVE

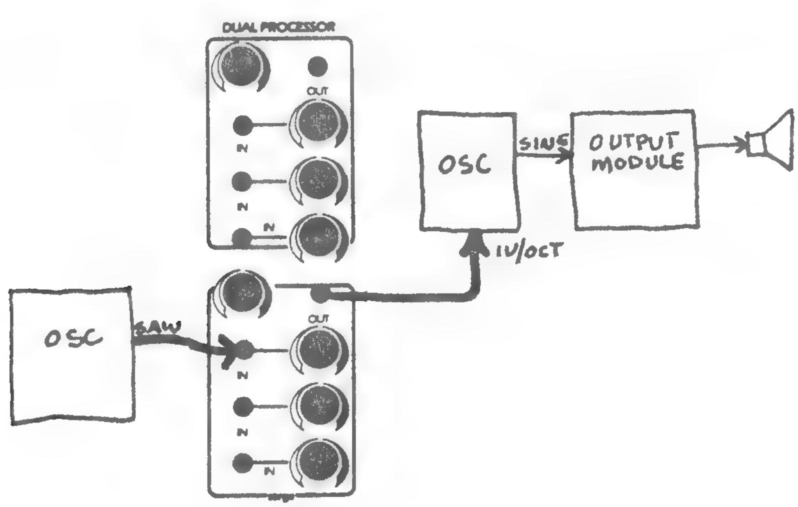


TRIANGLE WAVE MODULATING SINE WAVE

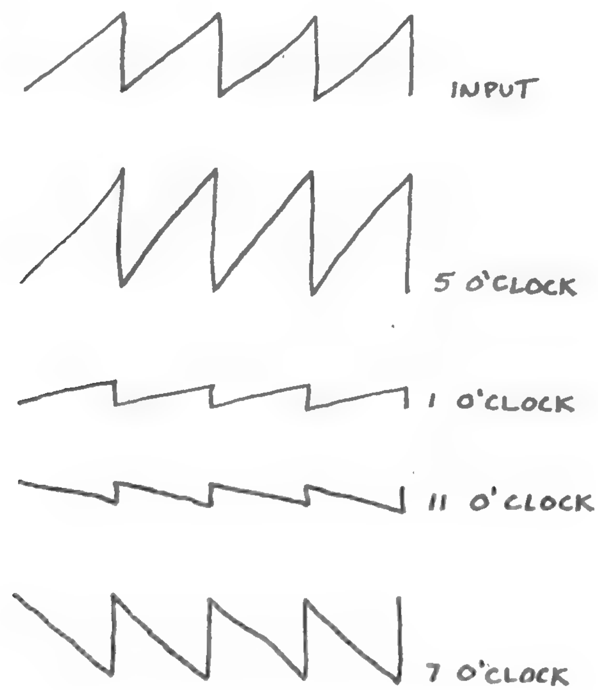
4.3 Increase the first oscillator's frequency slowly, listening carefully to the results. At first the sweeping will get faster and faster until a frequency approaching 20 Hertz is reached, at which point the sound takes on a multi-harmonic quality. This is called Frequency Modulation, or FM, because the frequency of the second oscillator is being changed or "modulated" at a rapid rate by the first. FM is a major technique of audio synthesis.

4.4 Set the FREQ of the first OSC so that the sweep of the second takes a few seconds. As the Pot associated with the control voltage input on the second OSC is moved from its full right position to a 12 o'clock setting, the sweeps will become shallower and shallower, although the time they take remains the same. As this Pot is turned to the left, the sweeps will have a greater and greater gain but an inverted one. Whereas a Pot set to the right causes the sweep to go upward and then suddenly fall downward, when it is set to the left the sweep is downward and the jump up. Control voltage inputs that have Pots of this type associated with them are called "Processed Inputs".

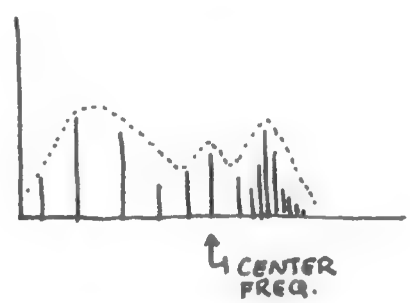
These Pots control a device internal to the module which can amplify, attenuate and/or invert a control voltage input. It is because of their extreme usefulness that they are they are the typical control voltage inputs of the Serge. The Serge also has Processor Modules which can be patched to serve the same function.



Below are some of the possible outputs of a Processor with a SAW input.



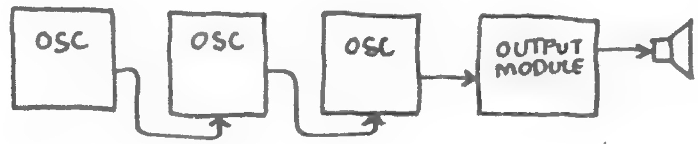
4.5 By setting the first OSC to HI (its range is switched to 20 to 20,000 Hertz, the audio range), an extremely wide range of sounds is possible with different combinations of FREQ and Processing pot settings on the two OSCs. This range can be extended even further by using different waveforms. The first OSC is referred to as the Modulator (or the signal, a term from radio broadcasting); the second OSC is referred to as the Modulated oscillator, or the Carrier. The setting of the processor, which determines the relative gain of the two OSCs, is called the Index. The frequency of the two oscillators and the setting of the Index determine the output of the modulated OSC. While the mathematics of FM is not simple, particularly with waveforms other than Sine waves, in general the spectrum of the output looks something like this:

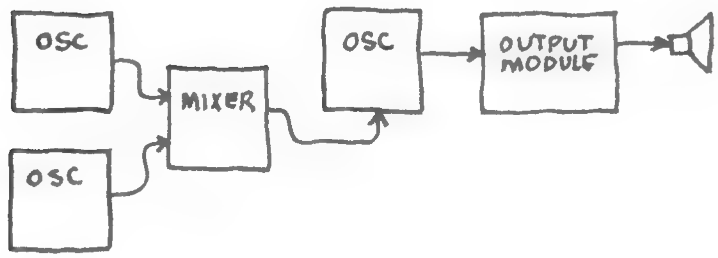


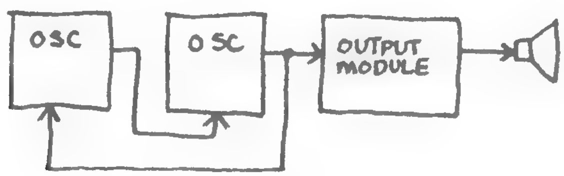
The frequency of the Modulated OSC sets the center frequency. There is an "overtone" or "undertone" every "f" Hertz where "f" is the frequency of the Modulator. The amplitude of these over/under-tones is determined by the Index and the frequencies of the oscillators. The overall shape of the amplitudes is butterfly and is called a Bessel function. In FM, sub-harmonics which would fall below 0 Hertz are "folded back" up to their "absolute" value. If the Modulated OSC is set at 200 Hertz and the Modulating OSC is at 60 Hertz then there should be sub-harmonics at 140, 80, 20, -40 and -100 Hertz. However these will be heard as 140, 100, 80, 40, and 20 Hertz.

Such mathematical descriptions, while interesting, are not vital to electronic music. Working with the synthesizer is rather like clay sculpture--you can work at the sound until it is right.

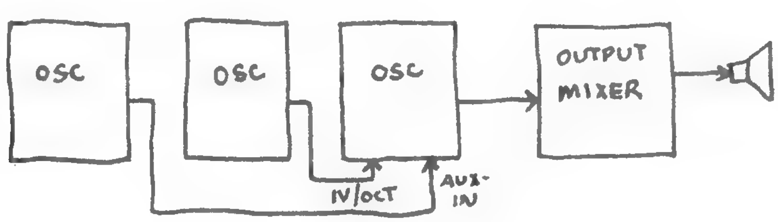
Keeping in mind that the output of an OSC, either modulated or unmodulated, is a varying voltage, and that such voltages can be used to control the frequency of other OSCs there are innumerable complex patches available to the synthesist.



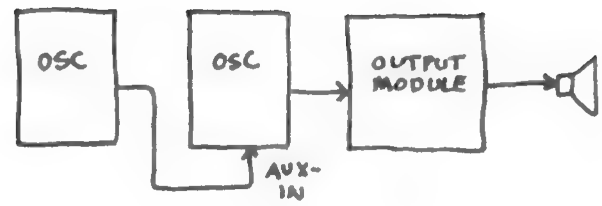




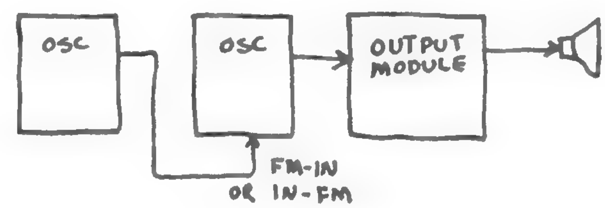
4.6 1V/OCT. The 1V/OCT control voltage on the oscillators is an extremely precise control voltage input whose effect is calibrated with detailed attention. The relationship of input voltage to output frequency is this: For every volt increase at the 1V/OCT input the OSC will rise EXACTLY one octave. One reason that such an input is valuable is that most synthesizer keyboards and other electronic music devices have output voltages that are set exactly to this relationship. Both the NTO and the PCO have two 1V/OCT inputs. (The second 1V/OCT input on the NTO is labelled Portamento In. It has another function associated with the pot and Control Voltage inputs below it. For now the pot should be turned full right.) When two different control voltages are received by an OSC they are added or summed together, after processing, so that both have an effect on the modulated oscillator and yet do not interact with each other.



4.7 FM-IN.



In the above patch, if both OSCs are set to audio frequencies, very interesting shifts in timbre occur when the Processing Pot is turned to different positions. However, when the AUX-IN is used, there are also apparent pitch changes.

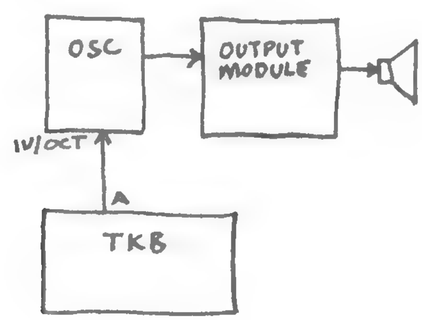


4.8 In this patch if the modulated OSC is a New Timbral Oscillator, then connect the modulating signal to the IN-FM. If the Precision VCO is used, use the FM-IN. Sweeping the associated pots of these inputs sets the Index. The sound produced should be similar to that produced by an audio voltage to the AUX-IN except that the over-all pitch does not seem to change as the INDEX is changed. The FM-IN and the IN-FM signal inputs are LINEAR, that is, equal rises of voltage produce equal increases in cycles per second.

STEP FIVE

All the modules in the previous step could have been found in a classical music studio except for the voltage controlled oscillator (although even it was found in some.) It is a powerful group of modules, with the oscillators providing the basic pitch material, the mixers adding these sounds together and adjusting their volumes, and the filters altering the timbre of the sound. Yet with only these modules many of the simplest sounds and patterns in music could not easily be achieved. In most musics there are discrete pitches whereas with the modules in the last step there were only sliding tones. Secondly it was hard to get non-repeating patterns.

The Touch Activated Keyboard Sequencer Module, or TKB, is a single large module designed specifically to produce control voltages. As discussed earlier, there is no physical or electrical difference between audio and control voltages other than that MOST audio voltages are between -2.5 volts and +2.5 volts, and all audio voltages are between 20 and 20,000 Hertz; while control voltages are between -12 and 0 volts, or 0 and +12, with frequencies anywhere between 0 and 500 Hertz. The actual difference between the two voltages are the uses to which they are put. The same voltage can be used in different ways. In one case it could be an audio voltage, in the other it could be a control voltage. However, some voltages are simply more useful in one situation than the other. The voltages produced by the TKB are designed to be used as control voltages.



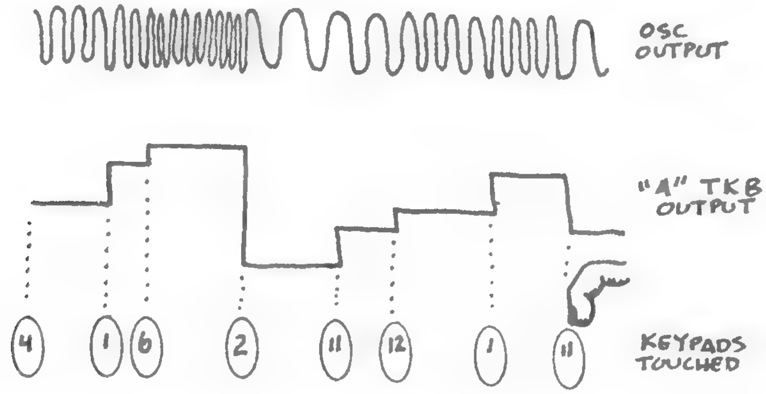
The TKB has four rows of pots across, labelled A,B,C, and D, and one row of keypads. There are 16 columns each with four pots (one from each row) and one keypad. At any given instant ONE and ONLY ONE column is activated and this is indicated by an LED (Light Emitting Diode) on the keypad of the respective column. These columns will be referred to from now on as STAGES.

5.1 The main outputs of the TKB are located at the top left-hand section on the module, enclosed in a border. There are five main voltage outputs (blue jacks) labelled A,B,C,D and ABCD. Patch the A output of the TKB to the 1V/OCT input of the OSC as shown in the above diagram. The OSC should be set to an audio frequency and its output sent to the output modules. Turn KEYS switch on and make sure that no other cords are patched to the TKB.

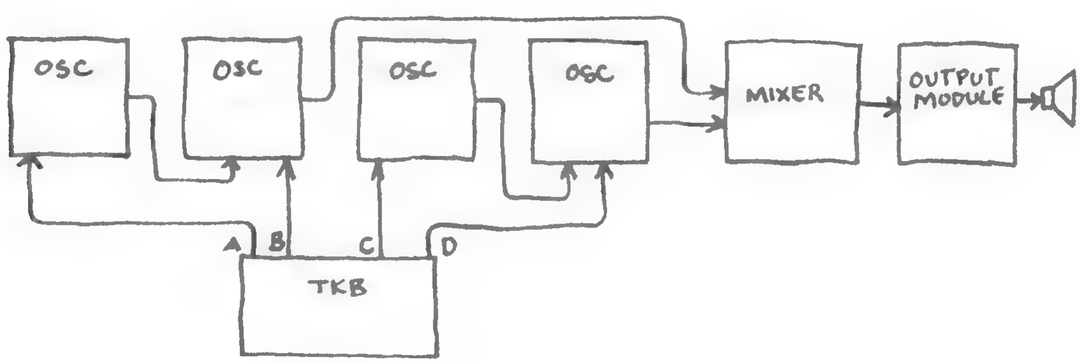
5.2 Touching keypad #1 activates stage #1 which is indicated by the LED that lights on keypad #1. Turn the pot in stage #1 and in row A (the top pot in stage #1) right and left. The OSC's frequency should shift up and down correspondingly. This pot is now remote-controlling the frequency of the OSC using a voltage that is appearing at output A.

5.3 Touch keypad #2 and set its A pot to a different setting then the A pot of stage #1. By alternately tapping keypads #1 and #2 you can get the OSC to produce two different "notes" or pitches without sliding from one to the other. This same procedure can be used to tune all 16 pots in row A. This is the tuneable keyboard.

The output of the TKB is NOT an audio voltage but rather a series of steady, or DC (direct current) voltages which are CONTROLLING the setting of the OSC (or whatever module or parameter the output is patched to). The OSC is designed to respond to these control voltages exactly like it responds to the turning of its pots. Just as the notes on a singer's score do not oscillate, so the voltages from the TKB do not oscillate but merely specify the OSC frequency. Below is a diagram of the voltage outputs of the TKB, and the OSC.

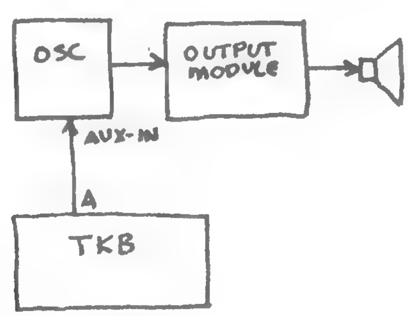


5.4 Patched in this fashion, none of the pots in Rows B,C or D have any effect. However, if it is re-patched so that the output of the TKB is taken from the B output instead of the A output then only the pots in Row B will be active. The same is true for Rows C and D. It is possible to use all four of these outputs (or as many as needed) SIMULTANEOUSLY as in the Patch below:

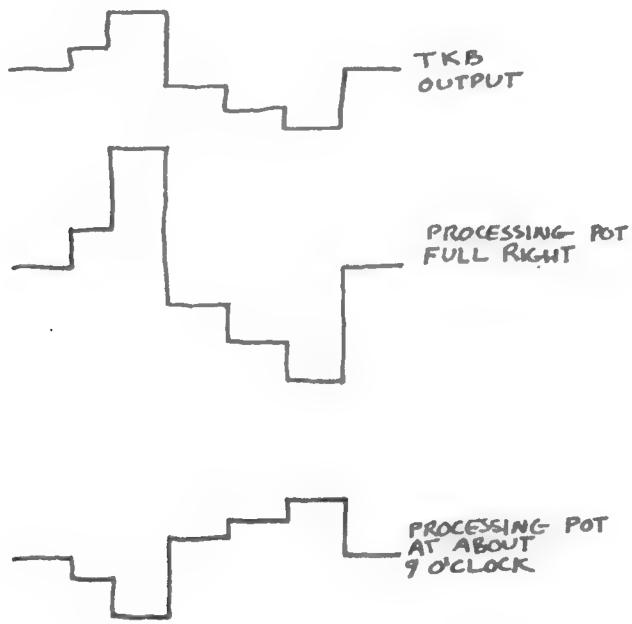


Now at each stage the pot in Row A controls the frequency of the Modulating OSC while the pot in Row B controls the base frequency of the modulated OSC. Row A and B could be replaced by any two rows. By touching the sixteen different keypads and setting the appropriate pots, sixteen different sounds can be set up and recalled in any order at the touch of a finger.

5.5

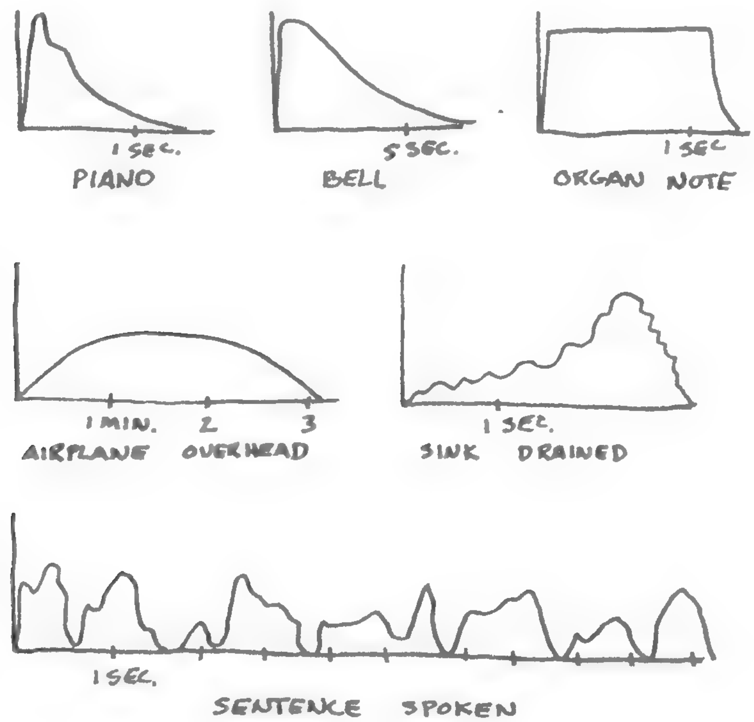


In the above patch the processing pot associated with the AUX-IN processes the incoming voltage from the TKB. Below are some typical processed TKB voltages:



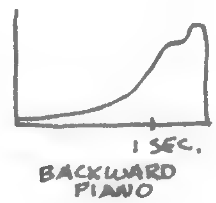
5.6 It is convenient to think of the TKB in this manner: All the pots in each row are tied to a common output (output A for the pots in row A for instance) but only one pot is activated and that is determined by which keypad was last touched. Since there are four rows, four parameters or modules can be controlled in 16 pre-set ways and these pre-sets, or stages, can be accessed directly by the touch pads.

When a piano note is struck, a bell gonged, a table tapped, an airplane flies overhead, a sentence spoken, a sink drained, an organ note sustained or when any other object makes a sound, that sound has an amplitude shape to it, an "envelope", that grows louder and softer in various ways as time passes.

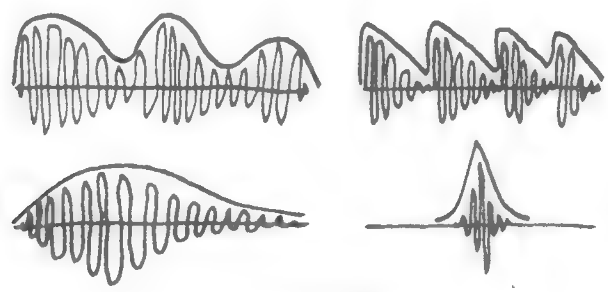


In these charts the loudness of the sounds is measured in db (decibels) while the duration is measured in units of time (seconds, minutes, etc.). The envelope refers only to the loudness of the sound, not to the frequency content of the sound. (We can again compare this "envelope to an envelope which holds a letter and think of the content of the sound as the letter.)

Every sound we hear has an envelope, and this is one of the ways various sounds are distinguished from one another. In this sense the envelope can be thought of as part of the timbre of a sound. Even artificial sounds have envelopes, for instance, this is the envelope of a piano, reversed:



AN ENVELOPE IS THE TRACE OF THE PEAK VOLTAGES OR PRESSURES OF A WAVE.



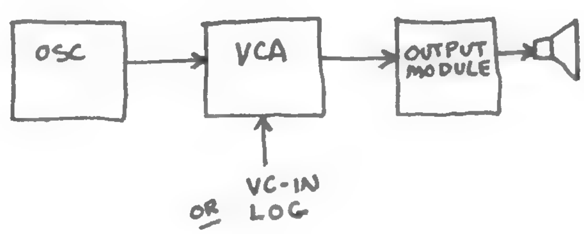
ENVELOPES

The Gain Pots on the mixer, when turned from left to right and then back to the left, can give the input signal an amplitude envelope. The device or module that automates this function is the Voltage Controlled Amplifier (VCA) or Gate and can be found both as an independent module and/or as part of most output modules. These modules are listed in STEP #6 of the first learning patch.

Every VCA has a signal input and a signal output and, like a mixer, has a Pot which can adjust the amplitude of the output in relation to the input without affecting any other parameter of the sound. In addition, the VCA has at least one Voltage Control input. As the control voltage rises, the amplitude of the output increases; as the control voltage falls so does the output amplitude. The control voltage, in effect, turns the Gain pot of the VCA by remote control.

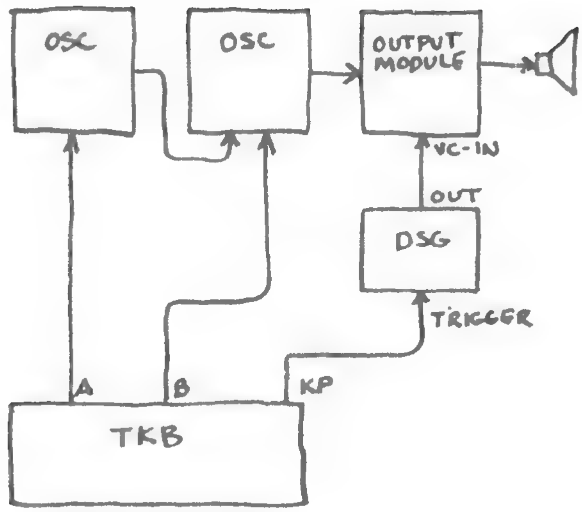
6.1 The Serge system has a wide range of VCAs, but for the purpose of this patch, the VCAs on the output module will be used. The signal input and output remain the same. Each input has an associated VC-in, usually located below or above the signal input (see the section on Output Mixers if your mixer is not mentioned in Learning Patch #1). Since Input #1 is used, VC-input #1 must be used to control its gain. Make sure there are no other patch cords connected to the module. The Gain associated with the input should be set to 10 o'clock. At this time, no sound should be heard from the VCA.

If the system being used has a separate VCA module, that module may be used instead of the VCAs on the output module in the following manner:



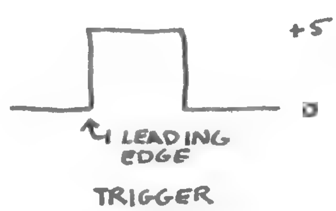
If the VCA is a dual or quad module make sure that the correct inputs, outputs and VC-ins are used.

6.2 Patch this patch:



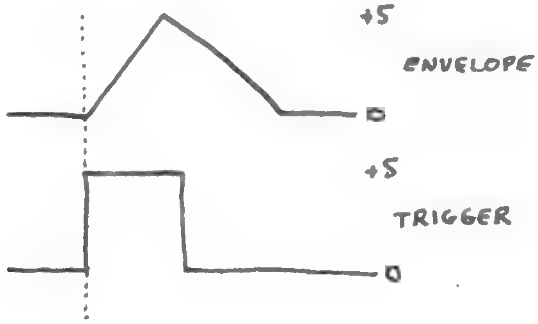
The Dual Universal Slope Generator contains two identical modules both of which will be referred to as a DSG. In these patches either one can be used.

The KP output of the TKB is in the same bank of outputs as the A and B outputs. It has a RED jack which indicates that it is a TRIGGER output which is the third kind of voltage on the system, Audio and Control being the other two. It's function is to either turn something on or turn something off. A Trigger voltage is always either 0 volts (its low state) or +5 volts (its high state). That moment when it goes from 0 volts to 5 volts is called its "positive transition" or "leading edge". It is this transition which turns functions on and off. All trigger outputs and all trigger inputs are RED jacks. It is possible in some cases to use appropriate control voltages to trigger certain modules, particularly if the control voltage has a sharp leading edge. (There are also a few places where these Trigger pulses can be used as audio waves if they are fast enough, or as control voltages if a two-level control voltage is desired.) If a control voltage can be thought of as turning a knob by remote control, a trigger is like pressing a button or tapping a key by remote control. A trigger looks like this:

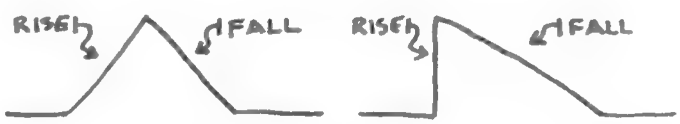


Each time a keypad is touched on the TKB, a trigger pulse appears at the KP (Key Pulse) output. It will remain in its HI state as long as the key is being touched.

In the above patch this trigger is sent to the TRIG-IN of the DSG at the bottom right hand of the module. When this module receives a Trigger pulse it produces exactly one VOLTAGE ENVELOPE.



This voltage envelope is a common, simple acoustic envelope similar to many musical envelopes such as piano, guitar, etc. It has two basic parts: The RISE and the FALL.



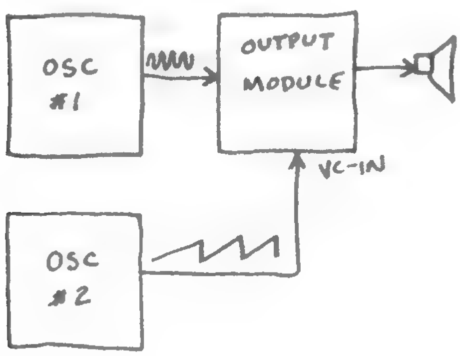
These two slopes are set by the two pots on the DSG labelled RISE and FALL. With these two pots the Rise and Fall time can be set anywhere from 1/1000th of a second to about 5 seconds.

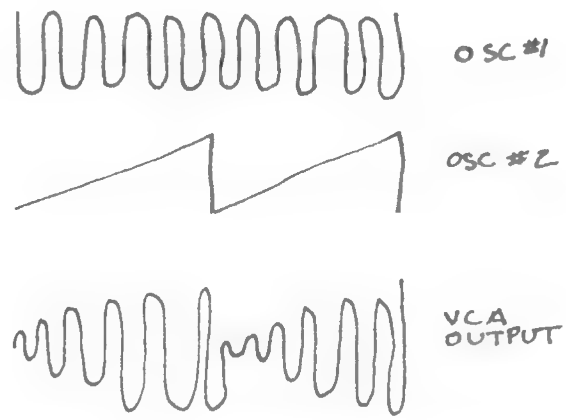
6.2 Patch from the KP out on the TKB to the TRIG-IN on the DSG and tap a keypad on the TKB. Directly above the output jack on the DSG is an LED (light emitting diode-- a small red light) whose brightness is proportional to the voltage of the envelope output. That is, as the envelope rises in voltage, the light gets brighter. Set the Rise and Fall pots to about 11 o'clock. The LED should take about one second to go from off to fully lit to off again. Different settings of the Rise and Fall pots will produce different timings. Set them so that they produce an envelope of this type:

6.3 Complete the patch from the output of the DSG to the VC-in of the VCA (the VC-controlled mixer will be referred to as a VCA when being used in that function). When doing so make sure that the Gain pot is set to the appropriate setting. If there is a small amount of sound "leaking" through, turn the Gain pot slowly to the left just until nothing is heard. Some VCA's (the UPAP, QMX, QCA, QVM, and SMX) can be overloaded if the initial gain is set too high. This will not damage the module, but it might overload your amplifier or speakers. Use caution on these modules, always starting out with the pot turned down, then increasing the gain with the control voltage applied until the sound is the proper level.

6.4 Touch a keypad on the TKB. This will cause a number of things to occur simultaneously. First, as already discussed, it will cause the Stage of the TKB that has been touched to be activated. At the same instant it causes a Trigger Pulse to be produced at the KP output of the TKB. This pulse Triggers the DSG to produce its envelope. This voltage envelope is patched to the VC-in of the VCA where the effect is as if turning the GAIN pot up and then down by remote control. As the voltage of the envelope increases the gain of the VCA increases. When the envelope starts its Fall, the gain of the VCA begins to decrease.

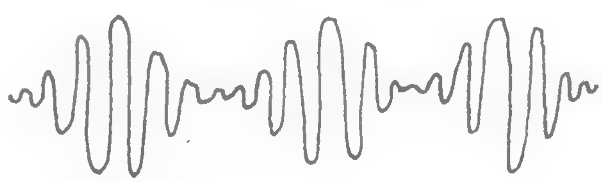
6.5 In general the voltage controllable parameters of a module are the same functions that can be controlled with its pots. For the VCA, then, the controllable function is its GAIN, where a High voltage to its VC-input creates a high GAIN and a low voltage produces a low GAIN. Once again, any voltage can be used to control the VCA, including an OSC. For instance



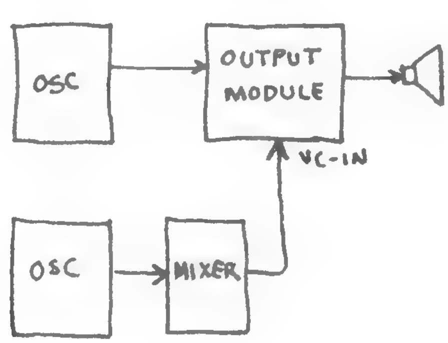


This will be heard, if the control OSC has a low enough frequency, as a kind of backwards sound which slowly gets louder and louder and suddenly cuts off -- only to begin again.

Using a SINE wave as the controlling voltage produces this effect:



6.6 Slowly increase the frequency of the modulating OSC. As you do so the sound will "beat" faster and faster. When this beating approaches 20 times per second (20 Hertz) a more complex sound appears that is somewhat similar to FM modulation. This sound is called Amplitude Modulation or AM. Like FM the sound is dependent on the Frequency of both OSCs and the relative amplitude between them. This relative amplitude, or Index, can be set in the following manner:

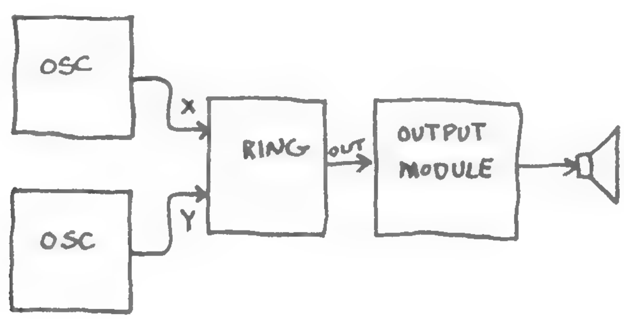


In these patches the GAIN on the mixer or VCA determines the Index. This technique can provide a wide range of sound types from tremelo to a hollow reedy sound to very complex sounds when non-SINE waves are used. In terms of the frequency spectrum, if a sine wave modulates a second sine wave, two NEW frequency components are produced, one being the sum and the other the difference between the two original sine waves. The original frequencies appear as well. That is, if the two original waves are 60 and 200 Hertz, then the output will be a mix of 60, 200, 260 and 140 Hertz waves.

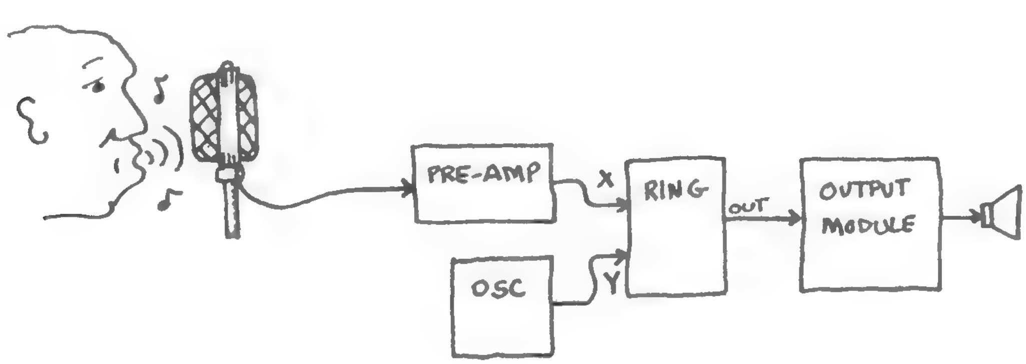
A module closely related to the VCA is the RING Modulator, which provides a third type of modulation along with FM and AM. Ring modulation is one of the oldest electronic music techniques and it is useful for producing complex and "odd" sounds similar to, but thicker than, the input sounds.

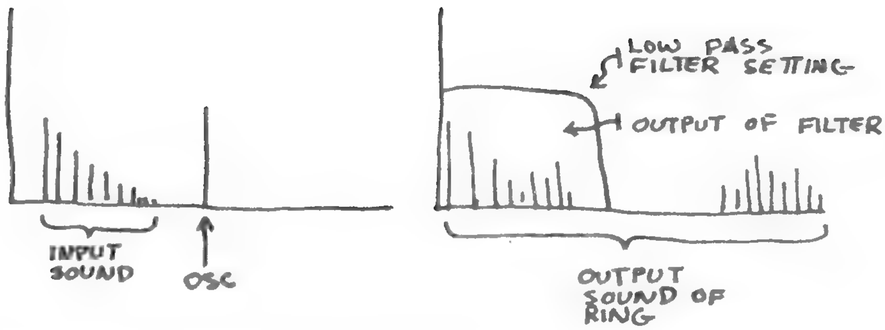
In its most basic mode a RING modulator takes two input frequencies and outputs the sum and difference frequencies ONLY. That is, if one input is 500 Hertz and the other is 160 Hertz, then the output is a 650 Hertz and a 350 Hertz wave mixed together. This differs from AM in that the original signals are cancelled out. If the input signals are complex, containing overtones, then every overtone of one wave is summed and differenced with every overtone of the second wave.

6.7 On Serge system RING modulators the two inputs are labelled X and Y (though in some ring modulators these inputs may be labelled as simply #1 and #2, or as signal and carrier). The output is labelled OUT. The pot at the bottom of the module is not a GAIN pot, but rather a pot which changes the function of the module from a standard VCA (full left) to a RING modulator when it is nearly full right.

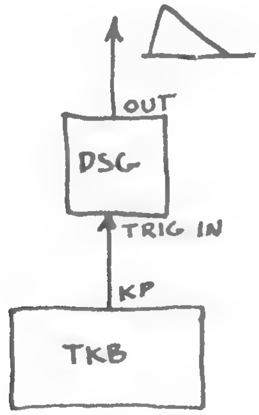


The RING modulator is often used in conjunction with sounds from the "real" world ("concrete" sounds) to give them an electronic feel. In this case, the "concrete" sound is fed to one input of the RING and the electronic sound to the other. It can also be used as a kind frequency shifter where a sound is shifted to a higher or lower frequency. For this, a filter must be used in conjunction with the ring modulator to filter out either the sum or difference component. This kind of frequency shifting alters significantly the harmonic relations of the overtones of the sound being shifted.



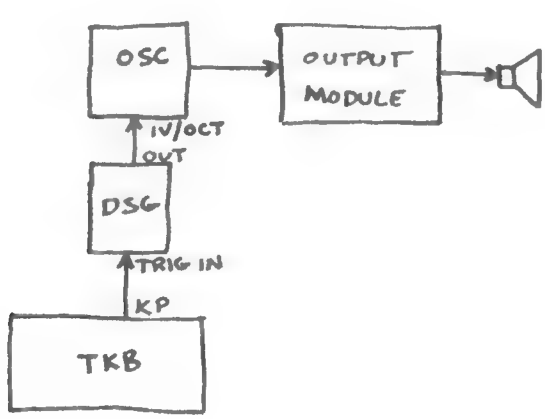


6.8 Some of the RING modulators on Serge systems have two auxiliary inputs, labeled VC-Y and VC-X, which act like VCAs for their respective input. They are useful for bringing out the original sound amidst the RING MODULATED sound.

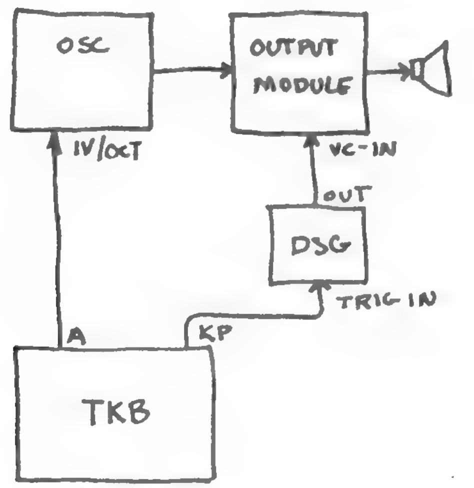


The DSG produces one voltage envelope every time it is Triggered. This trigger pulse may come from many places on the synthesizer but in the above patch came from the TKB each time a keypad was touched. The slope of the rise and fall of the envelope is set by the two pots labeled RISE and FALL.

6.9 While this voltage envelope is often used to control the amplitude or gain of a sound, it may be used to control any controllable module. In the following patch the envelope is controlling the frequency of an OSC:



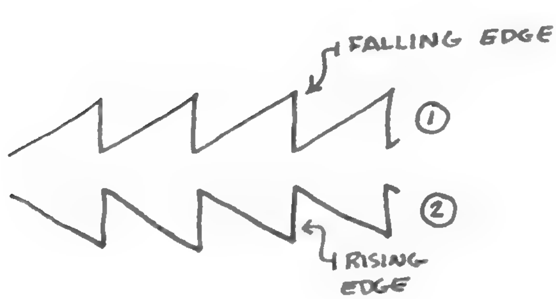
The following patch uses the TKB to trigger the DSG and control the frequency of the OSC.

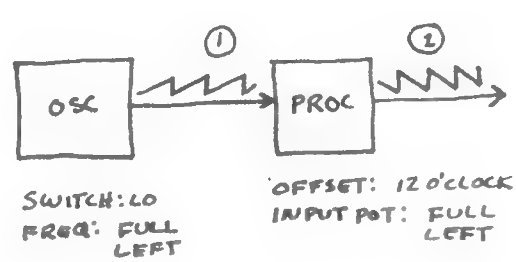


There are other features on the DSG and other ways of controlling it to extend its use far beyond this simple control function.

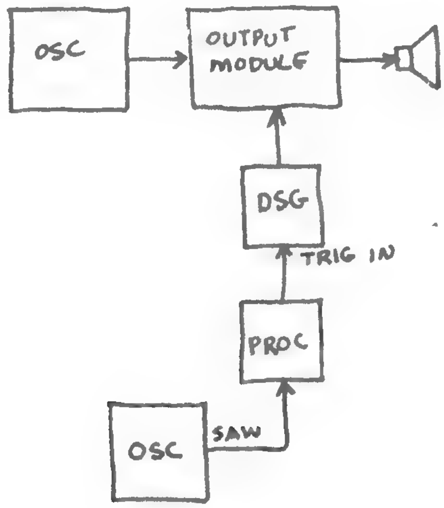
TRIGGERING FROM OTHER WAVEFORMS. The DSG, and in fact all TRIGGER-activated devices on the Serge, are triggered by the positive or rising edge of the Trigger pulse and not by the falling edge or the +5 voltage level itself. Not only a Trigger pulse from a trigger output but any sufficiently fast rising edge will trigger the DSG. The SAW wave output of PCO looks like this:

Note that it only has a Falling edge and therefore cannot trigger the DSG. However, this wave can be inverted by a Processor. The saw output of the OSC is patched into any of the three inputs of the PROC. The associated pot of the input should be set full left. Settings to the left of 12 o'clock on a processing input produce inverted outputs. While Processors usually accept control voltages, they can also accept audio voltages.





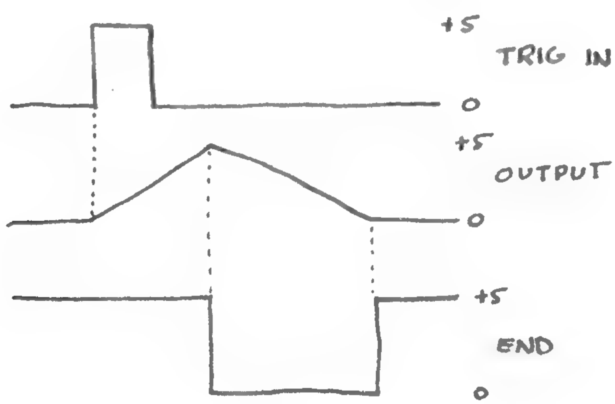
If the PCO is set LO and the frequency set very low (to the left), the output of the processor will be a series or "train" of rising edges that can be used to trigger the DSG over and over again.



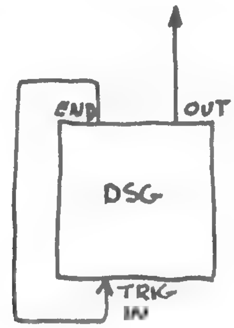
Some processors have an OFFSET pot that adds a set voltage to the output depending on its settting. If the processor being used in the above patch contains an offset pot it should be set at its 0 volt position (12 o'clock).

The DSG should be set so that the duration of the envelope as a whole is shorter than the "period" (the period of a wave is how long it takes to complete one cycle) of the OSC's sawtooth wave so that a full envelope can be generated before a new one is triggered. A DSG will not respond to a new trigger until it completes its entire Rise-Fall cycle.

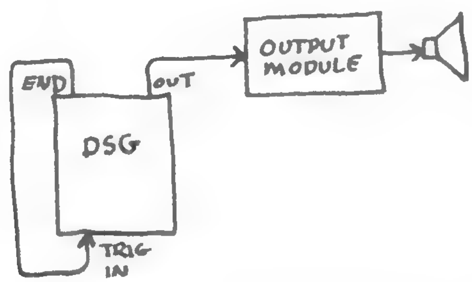
SELF-TRIGGERING and DELAY. The DSG has an END output that generates a rising edge at the completion of each envelope and REMAINS high until after another envelope is triggered.



This END Trigger can be used to Trigger any triggerable module, including ITSELF.

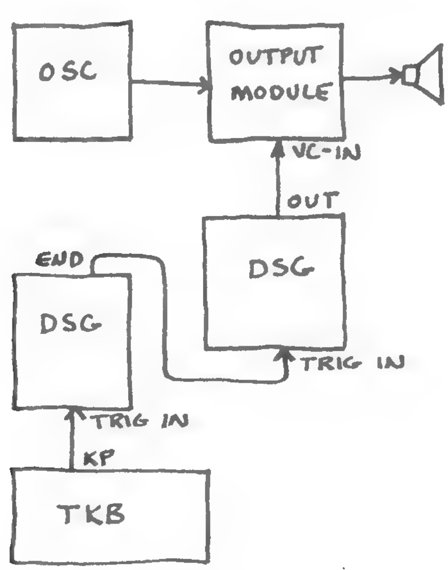


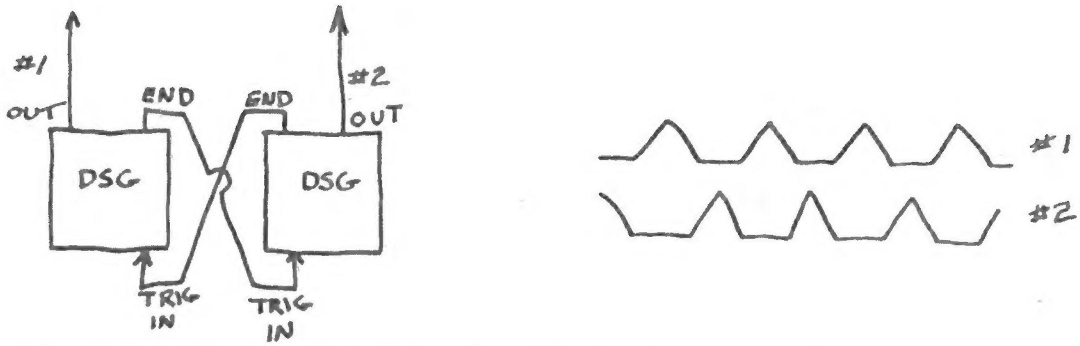
When the envelope has completed its cycle, a Trigger appears at the END jack. Since END is patched to TRIG-IN, the module is re-triggered and the cycle begins again. This patch turns the DSG into an oscillator! When the Rise and Fall times are set short enough, so that the total rise and fall time is less than one twentieth of a second, this OSC is within the audio range and can be heard directly through the speakers.



By adjusting the RISE and FALL pots different waveshapes can be achieved from saw to triangle.

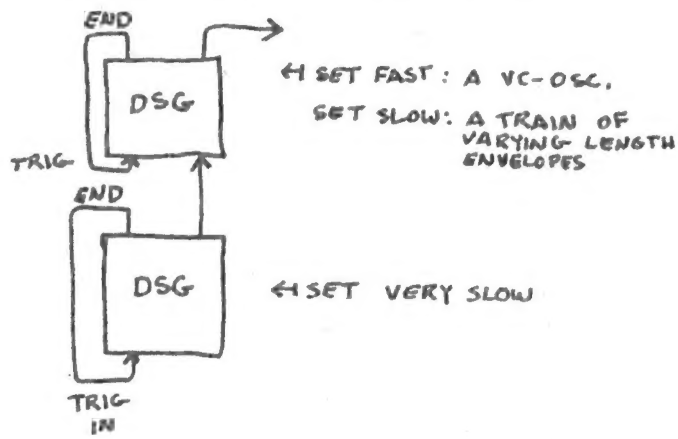
By using two DSGs a delay can be created between a trigger and the generation of an envelope or between successive envelopes of a DSG that is oscillating.



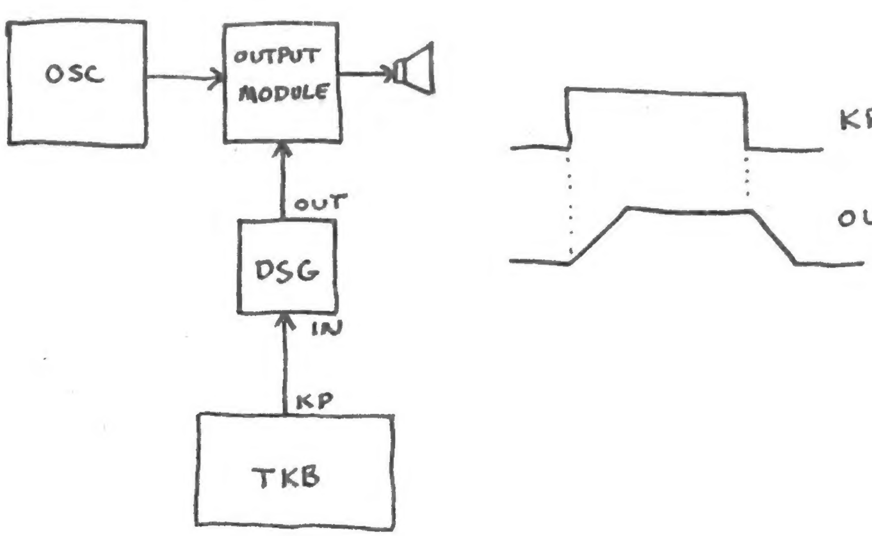


In these examples the length of the second envelope determines the delay. This envelope is not "heard" in any other way.

VOLTAGE CONTROL. Naturally the DSG can be voltage controlled itself. In this module different control voltages produce different Rise and Fall slopes and thus different length envelopes. The VC-IN has an associated 3-way switch which allows for 3 possible modes of control. When the switch is positioned to either RISE or FALL the control voltage controls EITHER the Rise or the Fall. In the center position a control voltage will control both Rise and Fall simultaneously. The VC-IN has an associated control voltage processor so that the control voltage can be amplified, attenuated and/or inverted. One good place to get control voltages to control one DSG is from another DSG:



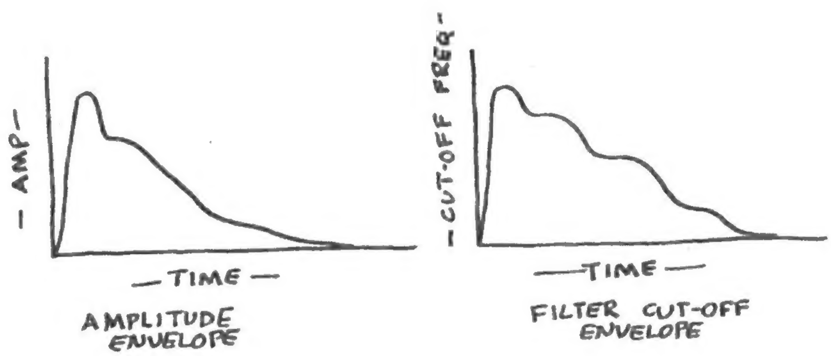
ENVELOPES WITH SUSTAIN. The DSG has an input labeled IN which accepts a voltage. If this voltage is higher or lower than the output voltage of the DSG, then the output voltage will rise or fall to the input voltage at a rate set by the RISE and FALL pots. This input is useful for making envelopes which sustain as long as the Trigger pulse remains high. For instance, the KP out on the TKB remains at 5 volts as long as a finger is held down on the keyboard.



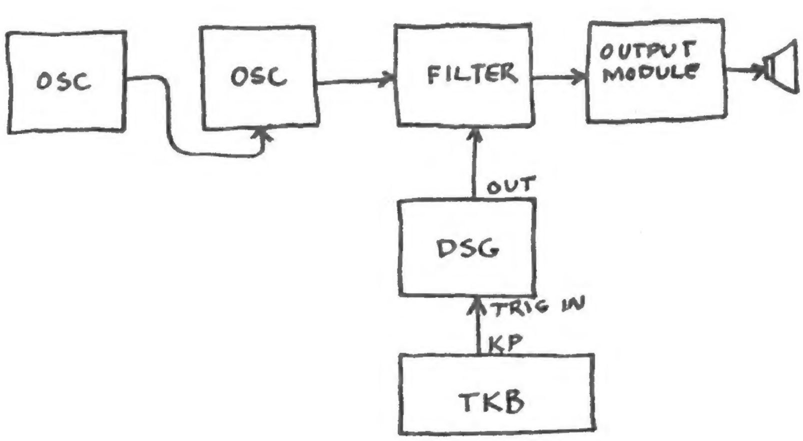
STEP SEVEN

When a piano note is sounded, not only does it have an overall amplitude envelope, but each harmonic or overtone has its own envelope. In most acoustic instruments the higher the frequency of the overtone the faster it dies away. The lowest tone, the fundamental, dies away last. This pattern is very much like closing down a fully opened LO PASS filter.

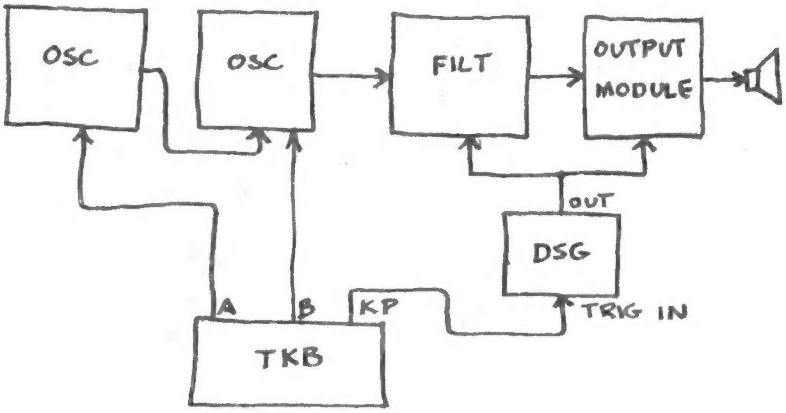
The control voltage applied to the filter sets the cut-off frequency. Usually the higher the voltage the higher the cut-off frequency. What makes it easier to simulate "natural" sounds using a VCA and a filter is that the amplitude envelope is often similar to the "harmonic spectrum envelope".



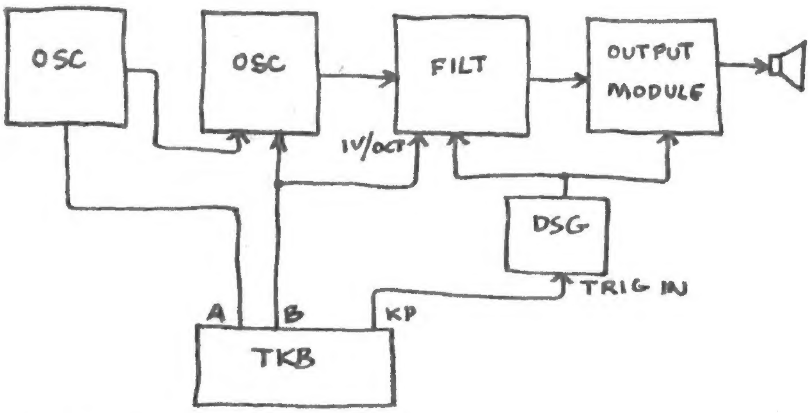
7.1 The similarities of these envelopes combined with the tautology "if all the harmonics die away the sound has died away", make it possible to simulate natural sound even without the use of a VCA. The cut-off frequency of the filter should be set low enough so that no sound gets through unless an envelope is applied. The envelope should be applied to the VCF input and the processing pot should be turned full right. The envelope should be set so that it rises rapidly and falls slowly. If the two OSCs are set to produce a fairly harmonic output, a bell-like sound should result.



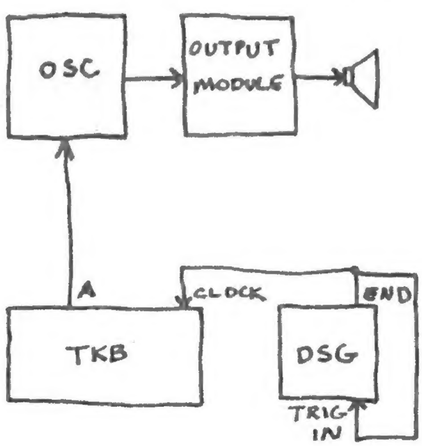
7.2 When this same patch is combined with a VCA controlled by the same envelope, and if the two OSCs are controlled by the A and B outputs of the TKB, the result can be an interesting keyboard instrument over which the composer has a lot of control.



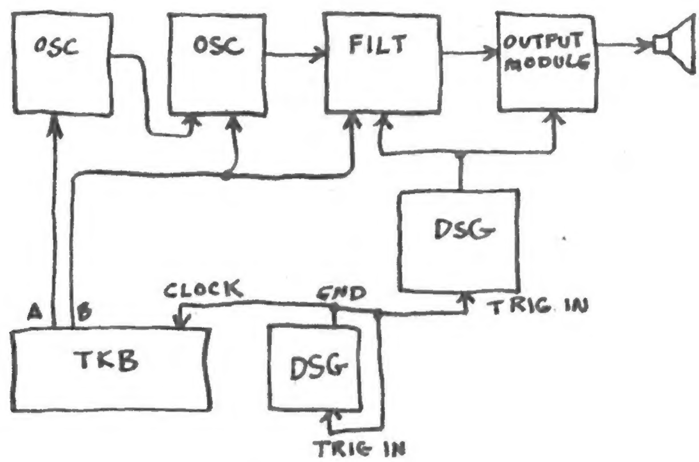
7.3 A limitation of this patch is that the initial cut-off frequency of the filter is always the same while the frequency of the OSC shifts under control of the TKB. A way to correct for this is by patching row B not only to OSC #2 but to the 1V/OCT input of the filter. Since these inputs are very precisely calibrated, and since they are being controlled by the same voltage, the filter and the OSC will "track", so that the cut-off frequency of the filter will follow the frequency of the oscillator.



7.4 This patch uses the TKB in only one of its two major modes: the keyboard mode. It is possible to use it in an automatic or SEQUENCER mode where different stages are accessed automatically. Near the upper right-hand corner of the TKB is a CLOCK input which accepts a trigger pulse (it is red, indicating a trigger in). Every time the TKB receives a trigger pulse at its CLOCK input it steps one stage to the right. If it is at stage 8 it will step to stage 9. If it is on stage 16, however, it "wraps around" to stage 1. Using a DSG set up as a slow OSC to provide trigger pulses, the following patch will step the TKB through its stages:



7.5 For this patch it is helpful to set all the pots in row A to different settings so that the different stages are distinguished from each other. Below is a logical extension of this automated TKB combined with the instrument sound we previously developed:



In this patch DSG #2 acts like a clock for the whole system. As it "ticks" it steps the TKB along and simultaneously triggers the #1 DSG.

7.6 A musical drawback with this patch is the regularity with which the system moves along. But since the DSG is voltage controllable, we have a way of altering the clock's rate by using a row of the TKB to Voltage Control it. The speed of the clock now has become an intregal part of the "musical instrument" that was constructed by patching together modules. By setting the pots in the row controlling the DSG, it is possible to set the time at each step-- in other words, to control the rhythm. Furthermore, row D can be used to control the length of DSG #1, the envelope to the VCA and to the Filter. With a thoughtful setting of the pots, 16 different sounds in a desired order, in any rhythm, can be produced and repeated: